Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) Publication number: **0 367 569 A2**

EUROPEAN PATENT APPLICATION

(21) Application number: 89311250.8

(22) Date of filing: 31.10.89

(51) Int. Cl.[5]: H04S 1/00 , H04R 27/00

(30) Priority: 31.10.88 JP 274726/88

(43) Date of publication of application: 09.05.90 Bulletin 90/19

(84) Designated Contracting States: DE FR GB

(71) Applicant: KABUSHIKI KAISHA TOSHIBA
72, Horikawa-Cho Saiwai-ku
Kawasaki-shi Kanagawa-ken(JP)

Applicant: TOSHIBA AUDIO VIDEO ENGINEERING CO. LTD.
3-9, Shinbashi 3-chome Minato-ku
Tokyo105(JP)

(72) Inventor: Sasaki, Akira Intellectual Property Division
Toshiba Corporation 1-1-1, Shinbaura
Minato-ku
Tokyo(JP)
Inventor: Suzuki, Katsuyoshi Intellectual Property Division
Toshiba Corporation 1-1-1, Shinbaura
Minato-ku
Tokyo(JP)
Inventor: Sakai, Kazuyasu Intellectual Property Division
Toshiba Corporation 1-1-1, Shinbaura
Minato-ku
Tokyo(JP)

(74) Representative: Muir, Ian R. et al
HASELTINE LAKE & CO. Hazlitt House 28 Southampton Buildings Chancery Lane
London WC2A 1AT(GB)

(54) Sound effect system.

(57) An audio signal processing apparatus for processing audio signal including an audio signal input circuit (116 -118) into which the audio signals are input, an audio signal analyzer (143) which analyzes the input audio signals and generates an output control signal, a sound effect processor (121) which performs a prescribed sound effect processing on the input audio signals and outputs a resulting audio signal, a controller (142) which controls the sound effect processor (121) to optimize the sound effect processing in response to the control signal from the audio signal analyzer (143) and an audio signal output circuit (122 -131) for outputting the resulting audio signal.

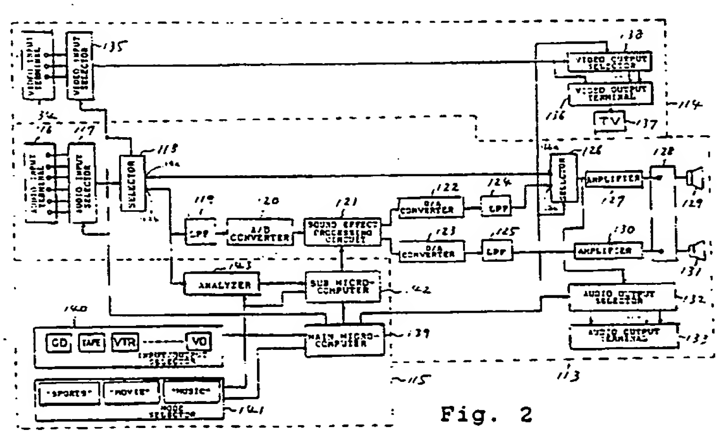


Fig. 2

EP 0 367 569 A2

SOUND EFFECT SYSTEM

The present invention relates generally to an audio signal processing apparatus, and more particularly to a sound effect system given by an audio signal processing apparatus which forms a sound field corresponding to an original sound source by applying sound effect processing to an audio signal.

Recently, many technical developments have been remarkably made in the field of audio equipment. For example, a stereophonic system has been widely used in audio equipment. A digital system also has been widely used for processing audio signals. These systems make the reproduced sound more similar to the original sound.

Furthermore, a sound effect processing apparatus capable of producing a specific reproduced sound field suitable to a listener's preference, by processing an audio source signal, such as music signal, has been strongly demanded in recent years.

FIGURE 1 shows a conventional audio signal processing apparatus for producing such a specific reproduced sound field. In FIGURE 1, an audio signal input terminal 101 receives an audio signal. The audio signal is supplied from a CD (Compact Disc) player, a tape player, VTR (Video Tape Player), LD (Laser Disc) player etc. The audio signal is applied to an analog to digital converter (referred to as A/D converter hereafter) 103 through a low pass filter (referred as to LPF hereafter) 102. The LPF 102 removes undesired high frequency components (referred as to HF or HF components) from the audio signal. The audio signal output from the LPF 102 is analog. The A/D converter 103 converts the analog audio signal to digital audio signal.

The digital signal is applied to a sound effect processor 104. The sound effect processor 104 produces a plurality of reverberation sound signals, e.g., two reverberation sound signals by processing the digital signal. The reverberation sound signals thus produced almost correspond to reverberation sounds in a concert hall or other sound fields. The sound effect processor 104 is typically constructed by, for example, delay units, adders, multipliers and the like.

The reverberation sound signals are converted into analog reverberation sound signals by digital to analog converters (referred as to D/A converters hereafter) 105 and 106. The analog reverberation sound signals are applied to amplifiers 109 and 110 through LPFs 107 and 108. The LPFS 107 and 108 remove undesired HF components from the analog reverberation sound signals. The amplifiers 109 and 110 amplify the reverberation sound signals and then supply the signals to loudspeakers 111 and 112.

Here, FIGURE 1 shows a one channel of the audio signal processing apparatus for the convenience's sake. However, the audio signal processing apparatus generally includes two channels for processing stereophonic related signals. Then, actually four sets of the loudspeakers are arranged at the front left and right and rear left and right. Thus, the loudspeakers gives a specific sound effect for listeners according to the reverberation sound signals.

In short, in this surround system, the sound effect processor 104 performs various signal processings for two channel input audio signals and by outputting four channel sounds, forms a sound field, surrounding listeners. As a result, listeners are able to listen as if they were actually in a concert hall or a sports arena.

When creating an atmosphere equivalent to, for instance, a concert hall, the sound effect processor 104 produces reverberation sound for 1 sec to 2 secs. However, this reverberation sound is produced not only for music but also when, for instance, an announcer or a master of ceremony (referred as to M.C. hereafter) is present. There is a problem with this because this reverberation sound is unnatural and it is hard to hear what the M.C. is saying.

Further, when creating a sound from a sports arena, the sound effect processor 104 produces, for instance, an echo of about several hundreds of milli-seconds (ms). This echo is produced not only for shouts of encouragement by the audience, but also is added to the voices of announcers or commentators and the same problems mentioned above are caused.

The present invention therefore seeks to provide an audio signal processing apparatus which is capable of creating an optimum sound effect according to the situation of sound source.

An audio signal processing apparatus according to one aspect of the present invention is provided with an audio signal input circuit into which the audio signals are input, an audio signal analysis circuit which analyzes the input audio signals and generates an output control signal, a sound effect processor which performs a prescribed sound effect processing on the input audio signals and outputs a resulting audio signal, a control circuit which controls the sound effect processor to optimize the sound effect processing in response to the control signal from the audio signal analysis circuit and an audio signal output circuit for outputting the resulting audio signal.

For a better understanding of the present in-

vention and many of the attendant advantages thereof, reference will be made by way of example to the accompanying drawings, wherein:

FIGURE 1 is a block diagram showing a construction of a conventional audio signal processing apparatus;

FIGURE 2 is a block diagram showing a first embodiment of the audio signal processing apparatus according to the present invention;

FIGURE 3 is a block diagram showing details of an audio signal analysis means of FIGURE 2;

FIGURE 4 is a block diagram showing details of a level adjuster of FIGURE 3;

FIGURE 5 is a block diagram showing another example of the level adjuster;

FIGURE 6 is a block diagram showing details of an LF level detector of FIGURE 3;

FIGURES 7 and 8 are frequency response charts of audio signals for explaining the operation of the LF level detector;

FIGURE 9 is a block diagram showing another example of the LF level detecter;

FIGURE 10 is a diagram showing details of an LF/HF level fluctuation detecter of FIGURE 3; FIGURES 11 to 14 are level diagrams of audio signals with reference to time for explaining the operations of the LF/HF level fluctuation detectors;

FIGURE 15 is a block diagram showing details of an L-R level detector of FIGURE 3;

FIGURES 16 and 17 are level diagrams of audio signals to time for explaining the operation of the L-R level detector;

FIGURE 18 is a block diagram showing another example of the LF/HF level fluctuation detector;

FIGURES 19 and 20 are frequency response charts of audio signals for explaining the operations of the LF/HF level fluctuation detectors of FIGURE 18;

FIGURES 21 and 22 are block diagrams showing modifications of the LF/HF level fluctuation detectors shown in FIGURE 18;

FIGURE 23 is a block diagram showing details of a detection signal processor of FIGURE 3;

FIGURE 24 is a waveform diagram for explaining the operation of the detection signal processor;

FIGURE 25 is a block diagram showing another example of the detection signal processor;

FIGURE 26 is a block diagram showing another construction of a gain adjuster;

FIGURE 27 is a block diagram showing another example of a frequency characteristic adjuster;

FIGURE 28 is a time chart for explaining the operation of the gain adjuster;

FIGURE 29 is a time chart for explaining the operation of a delay time adjuster;

FIGURE 30 is a schematic diagram showing details of a synchronizing circuit;

FIGURES 31 and 32 are time charts for explaining the operation of the synchronizing circuit;

FIGURE 33 is a block diagram showing another example of the synchronizing circuit;

FIGURE 34 is a time chart for explaining the operation of the synchronizing circuit of FIGURE 33;

FIGURE 35 is a schematic diagram showing still another example of the synchronizing circuit;

FIGURE 36 is a time chart for explaining the operation of the synchronizing circuit of FIGURE 35;

FIGURE 37 is a flow chart showing an operation of a main microcomputer of FIGURE 2;

FIGURE 38 is a block diagram showing a second embodiment of the audio signal processing apparatus according to the present invention; and

FIGURE 39 is a block diagram showing details of a video analyzer of FIGURE 38.

The present invention will be described in detail with reference to the FIGURES 2 through 39. Throughout drawings, reference numerals or letters used in FIGURE 1 will be used to designate like or equivalent elements for simplicity of explanation.

[First Embodiment]

FIGURE 2 is a block diagram showing the construction of the audio signal processing apparatus of the first embodiment of the present invention. The audio signal processing apparatus of the first embodiment is comprised of the audio system 113, video system 114 and control system 115. Further, in the drawing, a one channel audio system is presented as the audio system 113, but there may be two channel audio systems which operate together to form a stereophonic sound system.

Audio System 113

In the audio system 113, an audio signal input terminal block 116 is provided for receiving a plurality of audio signals from CD players, tape players, video players, LD (Laser Disc) players, etc. One of these audio signals input into the audio signal input terminal block 116 is selected by the audio input selector 117. The audio signal passed through the audio input selector 117 is further applied to a selector 118.

The selector 118 selects whether the audio signal is given a prescribed sound effect process-

ing or not, in cooperation with another selector 126. That is, the audio signal not to be given the sound effect processing is output from a first output terminal 118a of the selector 118. The audio signal selected for no processing is directly input to the selector 126, i.e., a first input terminal 126a of the selector 126. On the other hand, the audio signal to be given sound effect processing is output from a second output terminal 118b of the selector 118. The audio signal thus selected is input to a second input terminal 126b of the selector 126 through a sound effect processor as described in detail below.

The audio signal to be given the sound effect processing is applied to an A/D converter 120 through an LPF 119. The LPF 119 removes the high frequency components of the audio signal The A/D converter 120 converts the audio signal to a digital signal. The digital audio signal is input into a sound effect processor 121. The sound effect processor 121 produces a reverberation sound signal which resembles the reverberation sound in concert halls, stadiums etc. The digital audio signal and the reverberation sound signal are converted into analog signals by D/A converters 122 and 123, respectively. These analog signals are applied to LPFs 124 and 125. The LPFs 124 and 125 remove undesired high frequency components.

The analog audio signals output from the LPF 124 are applied to an amplifier 127 through the selector 126. The amplifier 127 amplifies the audio signals for driving loudspeakers 129 at the front side, which are connected through an output terminal block 128.

The analog audio signals output from the LPF 125 are applied to an amplifier 130. The amplifier 130 amplifies the audio signals for driving loudspeakers 131 at the rear side, which are connected through the output terminal block 128.

The audio signals not to be given the sound effect processing are applied to the amplifier 127 through only the selectors 118 and 126.

Further, the audio signal output from the selector 126 is applied to an additional audio output terminal block 133 through an audio output selector 132.

Video System 114

In the video system 114, a video signal input terminal block 134 is provided for receiving a plurality of video signals from CD players, video players, LD (Laser Disc) players, etc. One of these video signals input into the video signal input terminal block 134 is selected by the video input selector 135. The video signal passed through the video input selector 134 is supplied to a video display, e.g., a television receiver 137, through a video output terminal block 136 or both a video output selector 138 and a video output terminal block 136.

Control System 115

The control system 115 is provided with a main microcomputer 139, a sub microcomputer 142 and an analyzer 143 for controlling the audio system 113 and the video system 114.

The main microcomputer 139 controls the.audio input selector 117, the selectors 118 and 126, the audio output selector 132, the video input selector 135, and the video output selector 138 according to operation commands given by a user through an input/output selector 140. The input/output selector 140 is provided with a plurality of input source keys, e.g., "CD", "TAPE", "VTR", "LD" etc. These keys are operated by the user.

Further, the main microcomputer 139 controls the sound effect processor 121 through the sub microcomputer 142. The control of the sound effect processor 121 is made in response to the audio signal analysis means, i.e., an analyzer 143, and a mode selector 141 which is connected to the main microcomputer 139, as described in detail later. The mode selector 141 is provided with a plurality of mode keys, e.g., "SPORTS", "MOVIE", "MUSIC" etc. These keys are also operated by the user.

Then, the sub microcomputer 142 controls the sound effect processor 121 to optimize the operation thereof according to the signal.

Analyzer 143

FIGURE 3 shows the analyzer 143.

In FIGURE 3, the audio signal on the second output terminal 118b of the selector 118 (see FIGURE 2) is further applied to the analyzer 143. The audio signal is then input to the mode selection circuit 144. The mode selection circuit 144 sets up a mode of categories "SPORTS", "MOVIE" or "MUSIC". The mode setting operation in the mode selection circuit 144 is executed by the selected signal input through the mode selection key block 141. The audio signal passing through the mode selection circuit 144 is set at a fixed level by a level adjuster 145.

The audio signal set at the fixed level is applied to a level detector 146. The level detector 146 detects a level of a particular signal component of the audio signal for each mode, i. e., "SPORTS", "MOVIE" and "MUSIC". The particular component level detector block 146 is provided with a low frequency component (referred to as LF

or LF component hereafter) level detector 147, a low and high frequency components (referred to as LF HF or LF/HF components hereafter) level fluctuation detector 148, and an L-R signal (referred to as L-R or L-R signal hereafter) level detector 149.

If the "SPORTS" mode is selected, the audio signal is input into the LF level detecter 147. The LF level detector 147 detects the level of the LF component of the audio signal. If the "MOVIE" mode is selected, the audio signal is input into the LF/HF level fluctuation detector 148. The LF/HF level fluctuation detector 148 detects level fluctuations of the LF/HF components of the audio signal. If the "MUSIC" mode is selected, the audio signal is input into the L-R level detector 149. The L-R level detector 149 detects a level of the difference between two signals of the audio signals which are stereophonically related with each other.

The signal detected by the level detector 146 is output from the analyzer 143 through a detection signal processor 150. The detection signal processor 150 delays the following edge portion of the detected signal by a prescribed time constant.

The detected signal output from the analyzer 143 is applied to the sub microcomputer 142.

Level Adjuster 145

FIGURE 4 shows the level adjuster 145. The level adjuster 145 comprises a level detector 151 and an attenuator 152.

As shown in FIGURE 4, the audio signal is applied to both the level detector 151 and the attenuator 152 from the mode selector 144. The level detector 151 detects the level of the audio signal and then controls the attenuation of the attenuator 152 in response to the level. Thus, the level of the audio signal output from the attenuator 152 is maintained. Therefore, even when the level of the audio signal differs between the modes or audio signal sources, the sound source situation of the audio signal is always analyzed at the optimum state in the level detector 146.

FIGURE 5 shows another example of the level adjuster 145. The level adjuster 145 comprises a level detector 151 and an amplifier 153.

As shown in FIGURE 5, the audio signal is applied to the level detector 151 from the mode selector 144. The level detector 151 detects the level of the audio signal. The detected level is applied to the level detector 146 after being amplified by the amplifier 153. Thus, the level of the audio signal output from the attenuator 152 is kept constant. Therefore, even when the level of the audio signal differs among the modes or audio sources, the optimum level of the audio signal is always applied to the level detector 146 for analysis of the audio source situation.

Thus, the level adjusters 145 as shown in FIGURES 4 and adjust the level of the audio signal to a standard level signal which is suitable for the analysis of the audio signal in the level detector 146.

Level Detector 146

(1) LF Level Detector 147:

FIGURE 6 shows the LF level detector 147. The LF level detector 147 comprises an LPF 154, an integrator 155 and a comparator 156.

As shown in FIGURE 6, the audio signal output from the level adjuster 145 is applied to the LPF 154. The LPF 154 removes the desired HF components of the audio signal. The audio signal is then applied to the integrator 155 and is integrated. The integrated audio signal is applied to the comparator 156. The comparator 156 compares the audio signal with a reference level. The comparator 156 generates a detection signal when the level of the audio signal is higher than the reference level.

This LF level detector 147 is used in the "SPORTS" mode. In case of sports programs, the sound source situations are broadly divided into cheers or hand clapping and the voices of announcers or commentators. These situations differ from each other in their frequency. characteristic (spectrum). In the former situation, the LF thereof is relatively low as shown in FIGURE 7. On the other hand, in the latter situation, the LF thereof is relatively high, as shown in FIGURE 8.

The LF level detector 147 discriminates these sound sources from each other according to this frequency response characteristics, as shown in FIGURES 7 and 8. That is, the LF level detector 147 judges whether the audio signal has the characteristics of cheers or hand clapping or the characteristics of the voices of announcers or commentators from the level of the LF component of the audio signal. When the level of the LF component is higher than the reference level, it is judged that the voices of announcers or commentators is input to the audio signal processing apparatus. Then, the detection signal is output from the LF level detector 147.

FIGURE 9 shows another example of the LF level detector 147. This example of the LF level detector 147 further comprises a high pass filter (referred as to HPF hereafter) 159, another integrator 160 and a subtractor 161.

As shown in FIGURE 9, the LF component of the audio signal output from the level adjuster 145 is taken out by the LPF 157 and the integrator 158.

Further, the HF component of the audio signal is taken out by the HPF 159 and the integrator 160. These LF HF components of the audio signal are subtracted in the subtractor 161. The difference thereof is compared with the reference level. When the level of the difference signal is higher than the reference level, a detection signal is output from the comparator 162.

The LF level detector 147 of FIGURES 6 and 9 can be digitized. In this case, the audio signal is converted to digital signal before the application to the circuit.

(2) LF/HF Level Fluctuation Detector 148:

FIGURE 10 shows the LF/HF level fluctuation detector 148. The LF/HF level fluctuation detector 148 comprises an LPF 163, an HPF 165, a pair of integrators 164 and 166, a pair of capacitors 167 and 169, a pair of comparators 168 and 170 and an AND gate 171.

As shown in FIGURE 10, the LF component of the audio signal output from the level adjuster 145 is separated out by the LPF 163 and the integrator 164. The HF conponent of the audio signal is separated out by the HPF 165 and the integrator 166. DC components of the LF/HF components are removed by the capacitors 167 and 169. Thus, the AC components of the LF/HF components, i.e., the level fluctuations thereof, are compared with a reference level in the comparators 168 and 170, respectively. When level fluctuations of the low and high frequency components are higher than the reference levels, the comparators 168 and 170 output detection signals. These detection signals are applied to the AND gate 171. Thus, a detection signal of the LF/HF level fluctuation detector 148 is generated when both the detection signals of the comparators are simultaneously output, i.e., when both the level fluctuations of the LF/HF components of the audio signal are higher than the reference level.

The LF/HF level fluctuation detector 148 is used in the "MOVIE" mode. In case of movie programs, drama programs, etc., the sound source situations are broadly divided into narrations and others. These situations differ from each other in the level fluctuation of the audio signal. That is, in the case of narrations, the level fluctuations of the LF/HF components are relatively high, as shown in FIGURE 12. In the other case, e.g., cheers, the level of the HF component is high and its level fluctuation is small, as shown in FIGURE 11. In the case of the sound of waves, the levels of the LF/HF components are high but their fluctuations are small, as shown in FIGURE 13. In the case of the sound of cars, the level of the LF component only is high and its fluctuation is slightly large. The LF/HF level fluctuation detector 148 discriminates these sound source situations from each other according to their level fluctuation characteristics, as shown in FIGURES 11 to 14. That is, the LF/HF level fluctuation detector 148 judges whether the audio signal is a narration or something else in response to the level fluctuations of the LF/HF components of the audio signal. When both the level fluctuations of the LF/HF components are higher than the reference level, it is judged that a narration is input to the audio signal processing apparatus. Then, the detection signal is output from the LF/HF level fluctuation detector 148.

(3) L-R Level Detector 149:

FIGURE 15 shows the L-R level detector 149. The L-R level detector 149 comprises a subtractor 172, an integrator 173 and a comparator 174.

As shown in FIGURE 15, stereophonic signals L-ch and R- ch are subtracted from each other in the subtractor 172. Thus, the L-R signal between the stereophonic signals L-ch and R-ch is output from the subtractor 172. The L-R signal is integrated in the integrator 173. The integrated L-R signal is compared with a prescribed reference in the comparator 174. The comparator 174 outputs a detection signal when the level of this L-R signal is lower than the reference level.

The L-R level detector 149 is used in the "MUSIC" mode. In case of music programs, the audio signal may be broadly classified into two, i.e., the music performance and the voice of an M.C. These signals differ from each other in the stereophonic presence of the music performance and the voice of the M.C. The voice of the M.C. is close to the monaural state. That is, in case of the voice of M.C., the L-R signal is relatively low, as shown in FIGURE 16. On the other hand, in case of the music performance, the L-R signal is relatively high, as shown in FIGURE 17.

The L-R level detector 149 discriminates these sound source situations from each other according to the difference in stereophonic presence between the music performance and the voice of an M.C. That is, the L-R level detector 149 judges whether the audio signal is a music performance or the voice of an M.C. in response to the level of the L-R signal between stereophonic signals. When the L-R signal is lower than the reference level, it is judged that the voice of an M.C. is input to the audio signal processing apparatus. Then, the detection signal is output from the L-R level detector 149.

Each of the level detectors 146 is not limited only to those as referred above.

FIGURE 18 shows another example of the

LF HF level fluctuation detector 148. The LF·HF level fluctuation detector 148 comprises a band pass filter (referred as to BPF hereafter) 175, an HPF 177, a pair of integrators 176 and 178 and a subtractor 179.

As shown in FIGURE 18, the audio signal output from the level adjuster 145 is applied to both the BPF 175 and the HPF 177. The BPF 175 extracts the intermediate frequency component (referred as to IF or IF component herafter) of the audio signal. The IF component of the audio signal is integrated in the integrator 176. The HPF 177 extracts the HFcomponent of the audio signal. The HF component of the audio signal is integrated in the integrator 178. The integrated IF and HF signals are subtracted from each other in the subtractor 179. Thus, the difference of the component signals is output as the detection signal.

This LF/HF level fluctuation detector 148, as shown in FIGURE 19, is used in, for instance, the "MOVIE" mode. In case of movie programs, drama programs, etc., it may be desirable to divide the audio signal into words spoken indoors and words spoken outdoors. These signals differ from each other in frequency characteristic (spectrum). That is, the voices indoors have only IF components, as shown in FIGURE 19. On the other hand, the voices outdoors have HF noise in addition to the IF component in many cases, as shown in FIGURE 20. This circuit judges whether situations are indoor word situations or outdoor word situations according to the presence of the HF component in the audio signals in addition to the IF component.

FIGURE 21 shows a modification of the LF/HF level fluctuation detector 148 shown in FIGURE 18.

The LF/HF level fluctuation detector 148, as shown in FIGURE 21, compares the differential signal output from the subtractor 179 shown in FIGURE 18 with a standard signal level preset by the comparator 180, and outputs the detection signal as a binary number.

FIGURE 22 shows another modification of the LF/HF level fluctuation detector 148 shown in FIGURE 18. The LF/HF level fluctuation detector 148, as shown in FIGURE 22, is identical to that shown in FIGURE 18 with the exception of the HPF 177 which has been replaced with the LPF 181. This circuit is suitable for audio signals in an environment where LF noises such as cars, etc, are involved.

Further, in the examples only one situation detector is used for each mode. Needless to say, it is possible to combine multiple situation detectors with multiple modes. In this case, more accurate situation estimation can be expected.

Detection Signal Processor 150

FIGURE 23 is a diagram showing the construction of the detection signal processor 150.

As shown in FIGURE 23, the detection signal from the particular component level detector block 146 is delayed in its fall by the time constant circuit 182 which consists of resistors, capacitors, etc. As shown in FIGURE 24, the frequency of changes of the detection signal (FIGURE 24a) output from the level detector 146 is reduced, as shown in FIGURE 24b, by the time constant circuit 182, if the situation frequently changes. Thus, frequent changes of the detection signal from word to word are prevented and, as a result, any unnaturalness caused during listening is eliminated.

The detection signal processor 150 can be digitized by replacing the time constant circuit 182 with a delay circuit 183, as shown in FIGURE 25.

Sound Effect Processor 121

The sound effect processor 121 is generally composed of a sound field signal processor. The sound field signal processor comprises a gain adjuster, a delay time adjuster, a frequency characteristic adjuster and a phase adjuster. The sound effect processor can additionally include an IIR (Infinite Impulse Response) filter. The sound effect processor adjusts gain, delay time, frequency characteristic, and phase of the audio signal output from the A/D converter 120 under the control of the sub microcomputer 142 (see FIGURE 2).

Functions performed by the sound effect processor 121 are as follows:

The detection signal is input from the LF level detector 147, the LF/HF level fluctuation detector 148, or the L-R level detector 149 to the sub microcomputer 142 corresponding to a mode.

If the "SPORTS" mode is selected, the detection signal from the LF level detector 147 is input. Then, if it is judged that the audio signal source is voices of announcers or commentators, the adjustments shown below are carried out in the sound effect processor 121:

(1) The gain in the gain adjuster is reduced;

(2) The delay time is shortened by the delay time adjuster:

(3) The LF component is emphasized by the frequency characteristic adjuster; and

(4) The phase difference is reduced by the phase adjuster.

On the other hand, if is was judged from this detection signal that the sound source is cheers or hand clapping, the adjustments shown below are carried out in the sound effect processor 121:

(1) The gain in the gain adjuster is extended;

(2) The delay time is increased by the delay time adjuster;

(3) The emphasis of the LF component is reduced in the frequency characteristic adjuster; and

(4) The phase difference is increased by the phase adjuster.

If the "MOVIE" mode is selected, the detection signal from the LF HF level fluctuation detector 148 is input to the sound effect processor 121. Then, if it is judged from this detection signal that the sound source is voices, the adjustments shown below are carried out in the sound effect processor 121:

(1) The gain is reduced by the gain adjuster;

(2) The delay time is shortened by the delay time adjuster;

(3) The LF component is emphasizd by the frequency characteristic adjuster; and

(4) The phase difference of the audio signal is reduced by the phase adjuster.

On the other hand, if it is judged from this detection signal that the audio signal is other than words, the adjustments shown below are carried out in the sound effect processor 121:

(1) The gain is increased by the gain adjuster;

(2) The delay time is extended by the delay time adjuster;

(3) The emphasis of the LF component is reduced by the frequency characteristic adjuster; and

(4) The phase difference of the audio signal is increased by the phase adjuster.

If the "MUSIC" mode is selected, the detection signal from the L-R level detector 149 is input into the sound effect processor 121. Then, if it is judged from this detection signal that the sound source is the voices of the M.C., adjustments shown below are carried out in the sound effect processor 121:

(1) The gain is reduced by the gain adjuster;

(2) The delay time is shortened by the delay time adjuster;

(3) The LF component is emphasized by the frequency characteristic adjuster; and

(4) The phase difference of the audio signal is reduced by the phase adjuster.

On the other hand, if it is judged from this detection signal that the audio signal is performance such as singing, the adjustments shown below are carried out in the sound effect processor 121:

(1) The gain is increased by the gain adjuster;

(2) The delay time is extended by the delay time adjuster;

(3) The emphasis of the LF component is eliminated by the frequency characteristic adjuster; and

(4) The phase difference of the audio signal is increased by the phase adjuster.

Thus, the sound effect signal with optimum effect sound is generated in each mode according to the respective characteristics of the audio signals. For instance, the voices, etc., can be clearly reproduced. Inversely, cheers, songs, etc., can be joyfully listened to by listeners.

The gain adjuster, the delay time adjuster, the frequency characteristic adjuster and the phase adjuster can be provided independently from the sound effect processor 121. For instance, the gain adjuster may be an attenuator 184a, as shown in FIGURE 26. Further, the frequency characteristic adjuster may be a filter 184b, as shown in FIGURE 27.

Further, the sound effect in each mode, each of the gain, the delay time, the frequency characteristic and the phase can be changed in three ways or more.

Operation of Gain Adjuster

FIGURE 28 shows the timing charts for explaining the operation of the gain adjuster. In the gain adjuster, the gain adjusting signal is simply changed between two preset values (FIGURE 28b) in response to the detection signal (FIGURE 28a) from the analyzer 143. Thus, the reproduced sound effect is changed so that listeners hear the reproduced sound coming either from the center front or from all around.

There are various ways of the gain adjusting operation other than the above operation. For instance, the gain adjusting signal is changed with a prescribed delay time (FIGURE 28c). Thus, unnaturalness of the reproduced sound at the change is moderated. Another example is to change the gain adjusting signal with a prescribed hysteresis (FIGURE 28d). Thus, unnaturalness of the reproduced sound is also moderated. Further example is to change gradually the gain adjusting signal (FIGURE 28e). Thus, unnaturalness of the reproduced sound is further moderated. Still further example is to change the gain adjusting signal fast in case of voices spoken by announcers, etc., or slow in case of cheers or hand clapping (FIGURE 28f). Thus, an undesired reverberation is fast eliminated at the change to the voices of announcers, or a reverberation is gradually emphasized at the change to cheers or hand clapping.

Operation of Delay Time Adjuster

FIGURE 29 shows the timing charts for explaining the operation of the delay time adjuster.

As shown in FIGURE 29, the delay time adjusting signal is simply changed between two preset values (FIGURE 29b) in response to the detection signal (FIGURE 29a) from the analyzer 143. Thus, the reproduced sound effect is changed so that listeners hear the reproduced sound from the center front or from all around.

There are various ways of the delay time adjusting operation other than the above operation. For instance, the delay time adjusting signal is changed with a prescribed delay time (FIGURE 29d). Thus, unnaturalness of the reproduced sound at the change is moderated. Another example is to change the delay time adjusting signal with a prescribed hysteresis (FIGURE 29e). Thus, unnaturalness of the reproduced sound is also moderated. Further example is to change gradually the gain adjusting signal (FIGURE 29f). Thus, unnaturalness of the reproduced sound is further moderated. Still further example is to change the delay time adjusting signal fast in case of voices spoken by announcers, etc., or slow in case of cheers or hand clapping (FIGURE 29g). Thus, an undesired reverberation is fast eliminated at the change to the voices of announcers, or a reverberation is gradually emphasized at the change to cheers or hand clapping. The reverberation time can be changed (FIGURE 29h). As a result, it becomes possible to produce the optimum sound effect according to the detection signal.

Operation of Frequency Characteristic Adjuster

In the frequency characteristic adjuster, the LF component of the audio signal is increased or decreased according to the detection signal from the analyzer 143. Thus, the sound effect can be made conspicuous or inconspicuous for listeners.

There are various ways of the frequency characteristic adjusting operation other than the above operation. For instance, the gain of the HF component of the audio signal is adjusted in response to the detection signal from the analyzer 143. Another example is to eliminate the HF component of the audio signal in response to the detection signal. Further example is to eliminate the LF component of the audio signal in response to the detection signal. Still further example is to adjust the gain of the LF component of the center channel audio signal which does not include reverberation. Still further example is to adjust the frequency characteristic of the audio signal in response to the detection signal. In any of the above cases the sound effect can be made conspicuous or inconspicuous for listeners.

Operation of Phase Adjuster

In the phase adjuster, phases of specific left and right audio signals or phases of all signals are made to to be out of phase or in phase, according to the detection signal from the analyzer 143. Thus, it is possible to make the stereophonic sound effect strong or weak.

There are various ways of the phase adjusting operation other than the above operation. For instance, the phases of components of the audio signal are partially inverted in response to the detection signal. Thus, it is possible to change the stereophonic sound effects between the components of the audio signal.

Controls of Adjusters for Gain, Delay Time, Frequency Characteristic and Phase

This control operation is to be carried out by changing at least one parameter of the gain, the delay time, the frequency characteristic and the phase of the audio signal to preset values according to the detection signal from the analyzer 143. Thus, it is possible to produce an optimum sound effect.

There are various ways of the operations for changing the parameters other than the above operation. For instance, a prescribed parameter is changed with a delay time. Thus, unnaturalness of the reproduced sound at the change is moderated. Another example is to change a prescribed parameter with a hysteresis. Thus, unnaturalness of the reproduced sound at the change is also moderated. Further example is to change a prescribed parameter gradually in several steps. Still further example is to change a prescribed parameter fast in case of voices spoken by announcers, etc., or slow in case of cheers or hand clapping. Thus, an undesired reverberation is fast eliminated at the change to the voices of announcers, or a reverberation is gradually emphasized at the change to cheers or hand clapping.

Synchronizing Circuit in Sound Effect Processor 121

FIGURE 30 is a diagram showing the construction of a synchronizing circuit constituted in the sound effect processor 121. The synchronizing circuit comprises a decoder 185 and an edge detector 186.

In the decoder 185, a start pulse from the sound field signal processor is input into the terminal Res of the binary counter 187 and a clock synchronizing with the internal clock

(corresponding to 1 step) of the sound field signal processor is input into the terminal CK. A count data of the binary counter 187 is input to a count value setting circuit 188 which is comprised of an NAND gate, an inverter, etc., when a preset count data is detected. The preset count value responds to the timing when data read/write are not performed out in a RAM 193, which is described later.

In the edge detector 186, the control signal from the sub microcomputer 142 is input into the terminal D of the first flip-flop 189 and the decode output signal from the decoder 185 is input into the terminal CK via the inverter 190. The data signal from the first flip-flop 189 is input into the terminal D of the second flip-flop 191 and a decode signal output from the decoder 185 is input into the terminal CK. An inverted data signal output from the first flip-flop 189 and a data signal output from the second flip-flop 191 are supplied as write pulses to the sound effect processor 121 through the NAND gate.

FIGURE 31 shows a timing chart for explaining the operation of this synchronizing circuit. A start pulse output from the sound effect processor 121 is synchronizing with the clock "0" in synchronization with the internal clock of the sound effect processor 121.

When the start pulse is applied to the terminal Res of the binary counter 187 (FIGURE 31a), the binary counter 187 is reset. Starting from here, counting of clocks (from "0") input into the terminal CK of the binary counter 187 is commenced.

When the clocks are counted up to a set value, the decode signal is output from the count value setting circuit 188 (FIGURE 31b). When the control signal output from the sub microcomputer 142 has been input into the edge detector 186 (FIGURE 31c), a write pulse synchronized with the decode signal is output from the edge detector 186 (FIGURE 31d) and supplied to the sound effect processor 121.

This synchronizing circuit has the following effects:

In the sound effect processor 121, when audio signals are applied with the prescribed process (generation of effect sound, etc.), the control signals (gain data signal, delay time data signal, etc.) from the sub microcomputer 142 are input into its processor. In this processor, processes with dozens of steps per every sample of the audio signal are carried out based on the control signals, as shown in FIGURE 32.

Further, the sound effect processor 121 is provided with a sound effect processor 192, an RAM 193, etc., for holding one sample data of the audio signal prior and after the processing, in order to delay the audio signal, as shown in FIGURE 33. Thus, the write/read operations of the data for the RAM 193 are carried out for every step.

However, if the control signal from the sub microcomputer 142 is supplied to the sound effect processor 121 as the form of interruption (FIGURE 34b) during the processing (FIGURE 34a), as shown in FIGURE 34, the data in the RAM 193 are destroyed during this process. The destroys data causes noise.

The noise according to the data destruction can be prevented by taking the control signals from the sub microcomputer 142 into the sound effect processor 121 at the timing synchronizing with write pulse which is output from the synchronizing circuit as mentioned above. That is, at the timing when the data write/read are not carried out in the RAM 193.

Further, when the setting step is "0" or synchronization is simply needed, this circuit can be made in the simplified construction by omitting the decoder, as shown in FIGURE 35. The state of signals in this simplified construction is shown in FIGURE 36.

Operation of Sub Microcomputer 142

The sound effect varies for each mode. Now, an operation for gradually changing the sound will be explained in reference to FIGURE 37. FIGURE 37 shows a flow chart showing the operation of the sub microcomputer 142.

First, a prescribed initial step data N of an operation step data Ds is set for executing the sound effect processing. Then, a prescribed mode is set (Steps a - d). A prescribed control data Dc is set for every mode. The sub microcomputer 142 checks a detection signal Sd output from the analyzer 143 (Step e). If the detection signal Sd is present (Step f), a unit "1" of an operation step data Ds is subtracted from a current operation step data Dn of the operation step data Ds; i.e., Do = Do - 1 (Step g). This occurs in, e.g., the situation of voices spoken by announcers. Then, a following calculation is carried out with respect to a current control data Dc, a current step data Do of the operation step data Dn and the initial step data N (Step h):

$Dc = Dc \times (Do/N)$ (I)

The calculation result is supplied to the sound effect processor 121 as the new control data Dc. The sound effect processor 121 carries out to generate the sound effect in response to the new control data Dc.

If the detection signal is not present (Step f), the unit "1" is added to the current step data Do for advancing the operation step data Dc; i.e., Do = Do + 1 (Step i). This occurs in, e.g., the situation of cheers (Step i). Then, another calcula-

tion the same as the above calculation (I) is again carried out (Step h). The calculation result is supplied to the sound effect processor 121 as the new control data Dc.

If the mode is the same as before, the same operations are repeated (Steps j and k). Further, when the current operation step data Dc exceeds the preset initial data "N" (Step 1) or lowers below the unit data "1" (Step m), the operation is advanced without performing the above addition or the subtraction of the operation step data.

Further, if the mode has been changed (Steps j and k), the calculation result which was used in the mode previously executed is used as the initial control data of the new mode (Step n).

[Second Embodiment]

FIGURE 38 shows the construction of the audio signal processing apparatus according to the second embodiment of the present invention.

The audio signal processing apparatus shown in this diagram is provided with an analyzer 194 which uses not only audio signals but also video signals as materials for the audio signal analysis. FIGURE 39 shows details of the video signal analyzer which has been incorporated in the analyzer 194.

A video signal is applied to the analyzer 194 from the video input terminal 134 (see FIGURE 38). In FIGURE 39, a luminance signal of the video signal is input into a first BPF 195 in the analyzer 194. The first BPF 195 allows to pass therethrough the LF component of the luminance signal. The luminance signal is also input into a second BPF 196. The second BPF 196 allows to pass therethrough the HF component of the luminance signal. The LF/HF components of the luminance signal video signal output from the first and second BPFS 195 and 196 are detected as level signals by integrators 197 and 198, respectively. The level signals are compared with each other by a comparator 199.

Generally, video signals of a zoomed up subject are less in brightness and much even in color. On the other hand, video signals of subjects extending in a broad range showing various things are high in brightness and uneven in color. The video signal analyzer with the construction classifies the video signals by comparing the LF/HF components of the luminance signal. Thus, the audio signal processing apparatus shown in this embodiment changes the sound effect in response to the video signal analyzer.

The above embodiments of the present invention have been presented on the assumption that the audio system is a stereophonic sound system. But, it may be a monophonic sound system and in this case, the same effect in the above embodiments can be obtained.

As described above, according to the audio signal processing apparatus involved in the present invention, it is possible to produce optimum sound effect according to sound source situation at all times as the prescribed sound effect process is controlled to optimize it according to judged audio signal sound source situations.

As described above, the present invention can provide an extremely preferable sound effect system.

While there have been illustrated and described what are at present considered to be preferred embodiments of the present invention, it will be understood by those skilled in the art that various changes and modifications may be made, and equivalents may be substituted for elements thereof without departing from the true scope of the present invention. In addition, many modifications may be made to adapt a particular situation or material to the teaching of the present invention without departing from the central scope thereof. Therefore, it is intended that the present invention not be limited to the particular embodiment disclosed as the best mode contemplated for carrying out the present invention, but that the present invention include all embodiments falling within the scope of the appended claims.

The foregoing description and the drawings are regarded by the applicant as including a variety of individually inventive concepts, some of which may lie partially or wholly outside the scope of some or all of the following claims. The fact that the applicant has chosen at the time of filing of the present application to restrict the claimed scope of protection in accordance with the following claims is not to be taken as a disclaimer or alternative inventive concepts that are included in the contents of the application and could be defined by claims differing in scope from the following claims, which different claims may be adopted subsequently during prosecution, for example for the purposes of a divisional application.

Claims

1. An audio signal processing apparatus for processing audio signals, comprising an audio signal input means (116 -118) into which the audio signals are input, a sound effect processing means (121) which performs a prescribed sound effect processing on the input audio signals and outputs a resulting audio signal and an audio signal output means (122 - 131) for outputting the resulting audio signal,

CHARACTERIZED IN THAT the apparatus further comprises:
an audio signal analysis means (143) which analyzes the input audio signals and generates an output control signal; and
a control means (142) which controls the sound effect processing means (121) to optimize the sound effect processing in response to the control signal from the audio signal analysis means (143).

2. An audio signal processing apparatus as claimed in claim 1, wherein the audio signal analysis means (143) comprises:
a low frequency extracting means (154) which extracts low frequency signals from the audio signals; and
a signal level comparing means (156) which compares the level of the low frequency signals extracted by the low frequency extracting means (154) with a preset prescribed level and outputs the result of the comparison.

3. An audio signal processing apparatus as claimed in claim 1, wherein the audio signal analysis means (143) comprises:
a low frequency extracting means (157) which extracts low frequency signals from the audio signals;
a first signal level fluctuation determining means (158) which determines the fluctuating level of the low frequency signals extracted by the low frequency extracting means (157) and outputs a first level determining signal;
a high frequency component extracting means (159) which extracts high frequency component signals from the audio signals;
a second signal level fluctuation determining means (160) which determines the fluctuating level of the high frequency component signals extracted by the high frequency component extracting means (159) and outputs a second level determining signal; and
a signal level comparing means (162) which compares the first and second level determining signals and outputs the result of the comparison.

4. An audio signal processing apparatus as claimed in claim 1, wherein the audio signal analysis means (143) comprises:
an intermediate frequency component extracting means (175) which extracts intermediate frequency component signals from the audio signals;
a first signal level fluctuation determining means (176) which determines the fluctuating level of the intermediate frequency component signals extracted by the intermediate frequency extracting means (175) and outputs a first level fluctuation determining signal;
a high frequency component extracting means (177) which extracts high frequency component signals from the audio signals; and
a second signal level fluctuation determining means (178) which determines the fluctuating level of the high frequency component signals extracted by the high frequency component extracting means (177) and outputs a second level fluctuation determining signal; and
a signal level comparing means (179) which compares the first and second level fluctuation determining signals from the first and second signal level fluctuation determining means (176, 178) and outputs the result of the comparison.

5. An audio signal processing apparatus as claimed in claim 1, wherein the audio signal analysis means (143) comprises:
an intermediate frequency component extracting means (175) which extracts intermediate frequency component signals from the audio signals;
a first signal level fluctuation determining means (176) which determines the fluctuating level of the intermediate frequency component signals extracted by the intermediate frequency extracting means (175) and outputs a first level fluctuation determining signal;
a low frequency component extracting means (181) which extracts low frequency component signals from the audio signals; and
a second signal level fluctuation determining means (178) which determines the fluctuating level of the low frequency component signals extracted by the low frequency component extracting means (181) and outputs a second level fluctuation determining signal; and
a signal level comparing means (179) which compares the first and second level fluctuation determining signals from the first and second signal level fluctuation determining means (176, 181) and outputs the result of the comparison.

6. An audio signal processing apparatus as claimed in claim 1 wherein:
multiple channel audio signals are input independently into the audio signal processing means;
the audio signal analysis means (143) is provided with a signal level difference determining means (149) which determines the difference in signal level between the multiple channel audio signals and a signal level comparing means (174) which compares the determined signal level difference with a preset prescribed level and outputs the result of the comparison; and
the sound effect processing means (121) performs the sound effect processing on the multiple channel audio signals in response to the output of the signal level comparing means (174).

7. An audio signal processing apparatus as claimed in any preceding claim wherein the sound effect processing means (121, 184) adjusts the gain of the audio signals.

8. An audio signal processing apparatus as claimed in claim 7 wherein the sound effect pro-

cessing means (121, 184, 182) changes the gain of the audio signals gradually.

9. An audio signal processing apparatus as claimed in any preceding claim wherein the sound effect processing means (121, 183) adjusts the delay time of the audio signals.

10. An audio signal processing apparatus as claimed in claim 9 wherein the sound effect processing means (121, 183, 182) changes the delay time of the audio signals gradually.

11. An audio signal processing apparatus as claimed in claim 9 wherein the sound effect processing means (121, 183) adjusts the delay time of the audio signals to provide either a long or a short reverberation time of the audio signal.

12. An audio signal processing apparatus as claimed in any preceding claim wherein the sound effect processing means (121, 185) adjusts the frequency characteristic of the audio signals.

13. An audio signal processing apparatus as claimed in claim 12 wherein the sound effect processing means (121, 185) adjusts the frequency characteristic of the audio signals by dividing the audio signal into low frequency signal and high frequency component signal and adjusting the gain of either or both of the low and high frequency component signals.

14. An audio signal processing apparatus as claimed in any preceding claim wherein the sound effect processing means (121) adjusts the phase of the audio signals.

15. An audio signal processing apparatus as claimed in claim 14 wherein the sound effect processing means (121) adjusts the phase of the audio signals on multiple channels.

16. An audio signal processing apparatus as claimed in any preceding claim wherein the sound effect processing means (121) adjusts one or more of the gain, delay time, frequency characteristic, and phase of the input audio signals.

17. An audio signal processing apparatus as claimed in any preceding claim, comprising:
a signal level detecting means (151) which detects the level of the audio signal; and
a signal level control means (152) which controls the signal level of the audio signal in response to the level detected by the signal level detecting means (151).

18. An audio signal processing apparatus as claimed in any preceding claim,
wherein the audio signal analysis means (143) comprises a delay means (183) which acts to delay the output control signal.

19. An audio signal processing apparatus for processing audio signal comprising an audio signal input means (116 -118) into which the audio signals are input, a video signal input means (134, 135) into which video signals are input, a sound effect processing means (121) which performs a prescribed sound effect processing on the input audio signals and outputs a resulting audio signal and an audio signal output means (122 - 131) for outputting the resulting audio signal, CHARACTERIZED IN THAT, the apparatus further comprises:
a video signal analysis means (143) which analyzes the input video signals and generates an output control signal; and
a control means (142) which controls the sound effect processing means (121) to optimize the sound effect processing in response to the control signal from the video signal analysis means (143).

20. An audio signal processing apparatus as claimed in claim 19, wherein the video signal analysis means (143) comprises:
a low frequency extracting means (195) which extracts low frequency signals from the luminance signal contained in the video signals;
a first signal level determining means (197) which determines the level of the low frequency signals extracted by the low frequency extracting means (195) and outputs a first level determining signal;
a high frequency component extracting means (196) which extracts high frequency component signals from the luminance signal and outputs a second level determining signal;
a second signal level determining means (198) which determines the level of the high frequency component signals extracted by the high frequency component extracting means (196) and outputs a second level determining signal; and
a signal level comparing means (199) which compares the first and second level determining signals and outputs the result of the comparison.

21. An audio signal processing apparatus for processing audio signal comprising an audio signal input means (116 -118) into which the audio signals are input, a video signal input means (134, 135) into video signals are input, a sound effect processing means (121) which performs a prescribed sound effect processing on the input audio signals and outputs a resulting audio signal and an audio signal output means (122 - 131) for outputting the resulting audio signal, CHARACTERIZED IN THAT the apparatus further comprises:
an audio signal analysis means (143) which analyzes the input audio signals and generates a first output control signal;
a video signal analysis means (143) which analyzes the input video signals and generates a second output control signal; and
a control means (142) which controls the sound effect processing means (121) to optimize the sound effect processing in response to the first and second control signals from the audio and video signal analysis means (121).

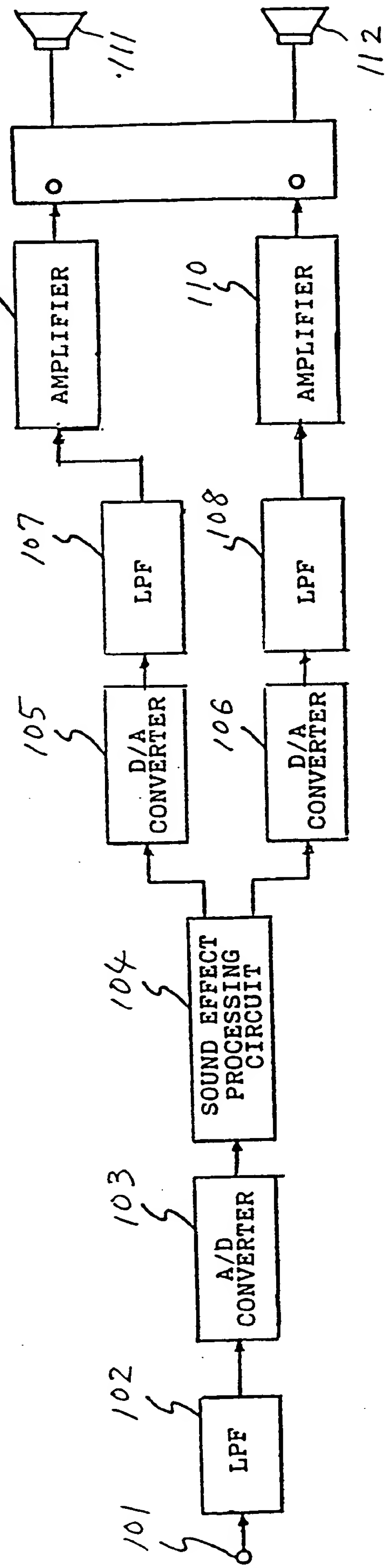


Fig. 1
(PRIOR ART)

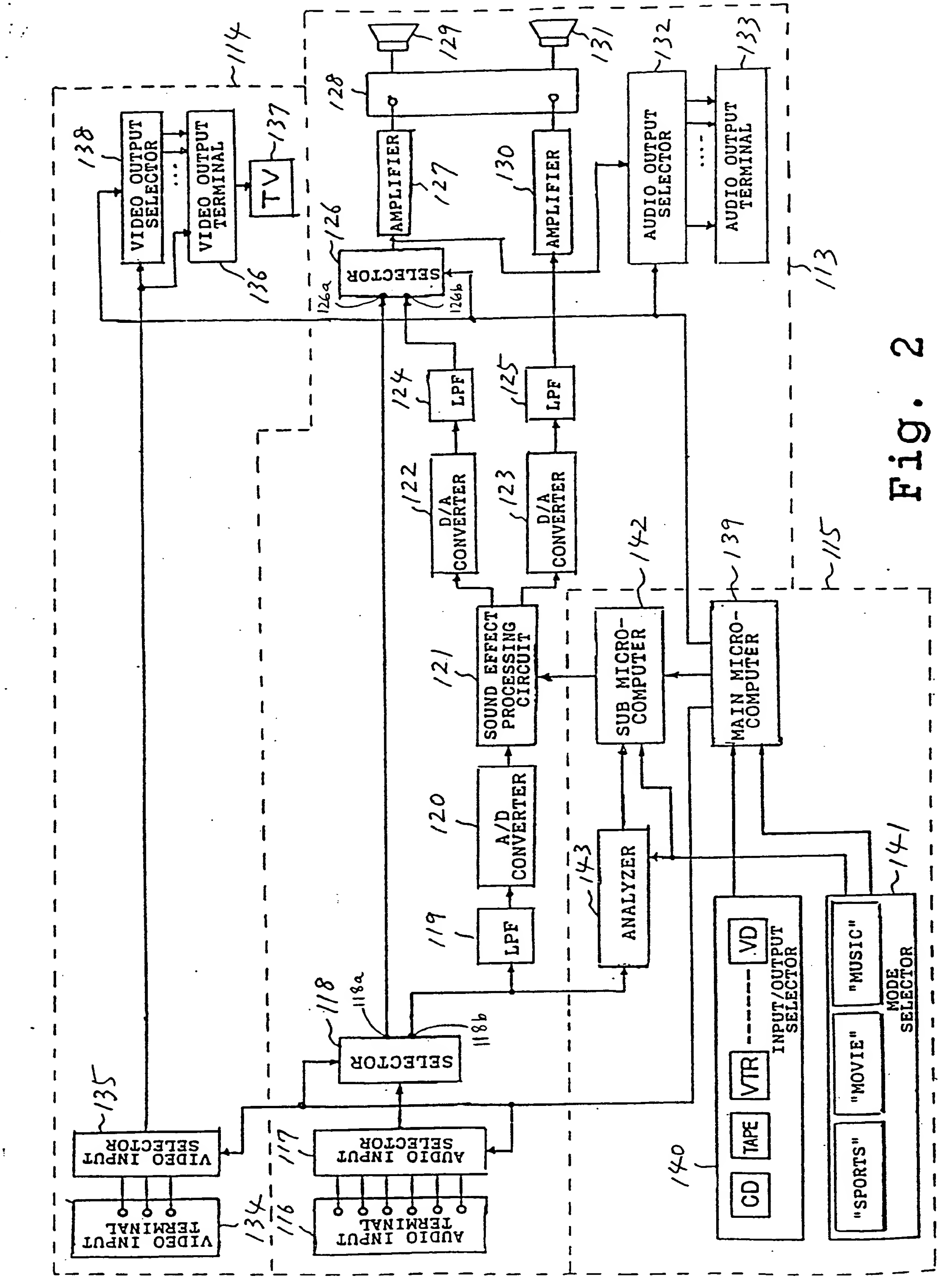

114
113
115
VIDEO OUTPUT SELECTOR
138
VIDEO OUTPUT TERMINAL
136
TV
137
128
129
131
AMPLIFIER
127
AMPLIFIER
130
AUDIO OUTPUT SELECTOR
132
AUDIO OUTPUT TERMINAL
133
SELECTOR
126
126a
126b
LPF
124
LPF
125
D/A CONVERTER
122
D/A CONVERTER
123
SOUND EFFECT PROCESSING CIRCUIT
121
SUB MICRO-COMPUTER
142
MAIN MICRO-COMPUTER
139
A/D CONVERTER
120
ANALYZER
143
LPF
119
SELECTOR
118
118a
118b
VIDEO INPUT SELECTOR
135
AUDIO INPUT SELECTOR
117
VIDEO INPUT TERMINAL
134
AUDIO INPUT TERMINAL
116
CD
TAPE
VTR
VD
INPUT/OUTPUT SELECTOR
140
"SPORTS"
"MOVIE"
"MUSIC"
MODE SELECTOR
141

Fig. 2

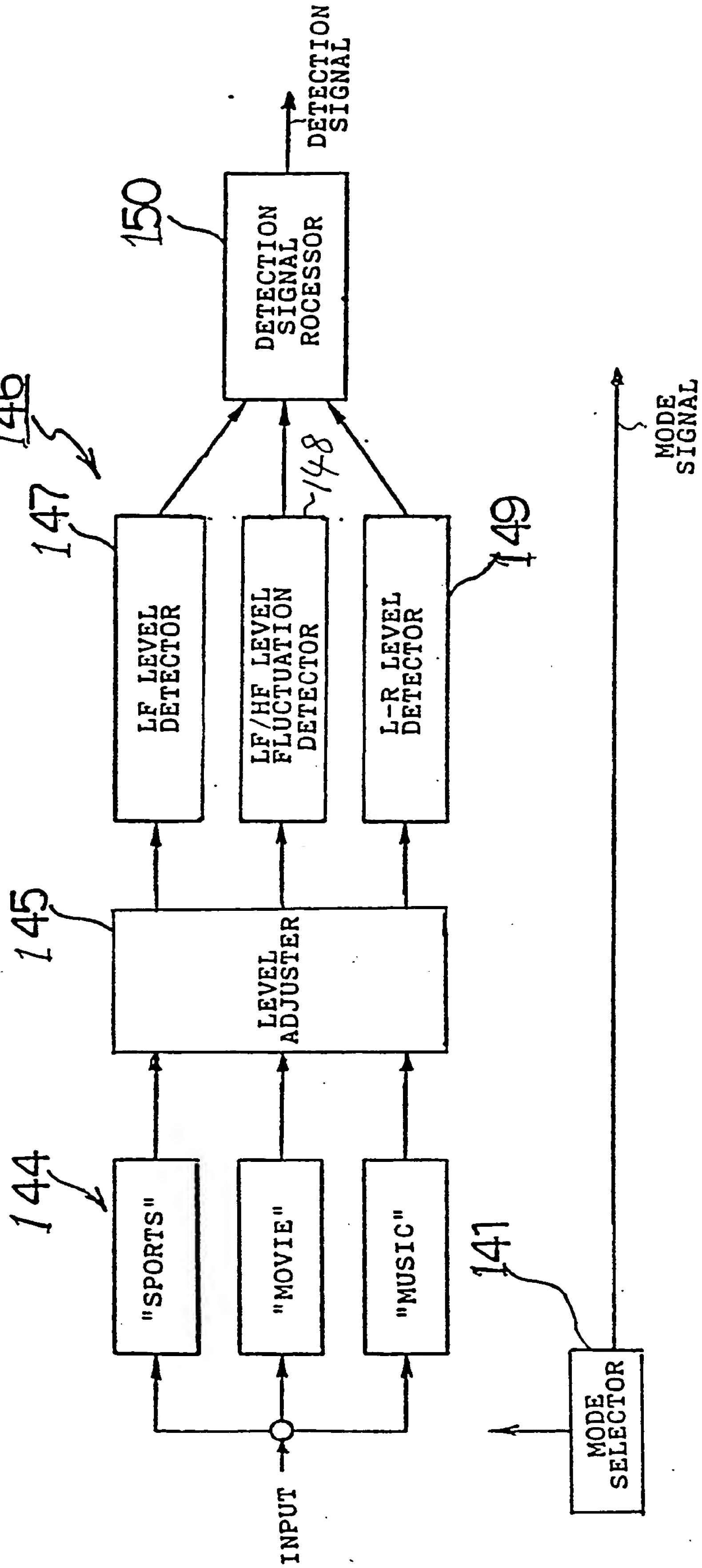

INPUT
"SPORTS"
"MOVIE"
"MUSIC"
144
141
MODE SELECTOR
LEVEL ADJUSTER
145
LF LEVEL DETECTOR
LF/HF LEVEL FLUCTUATION DETECTOR
L-R LEVEL DETECTOR
147
148
149
146
DETECTION SIGNAL ROCESSOR
150
DETECTION SIGNAL
MODE SIGNAL

Fig. 3

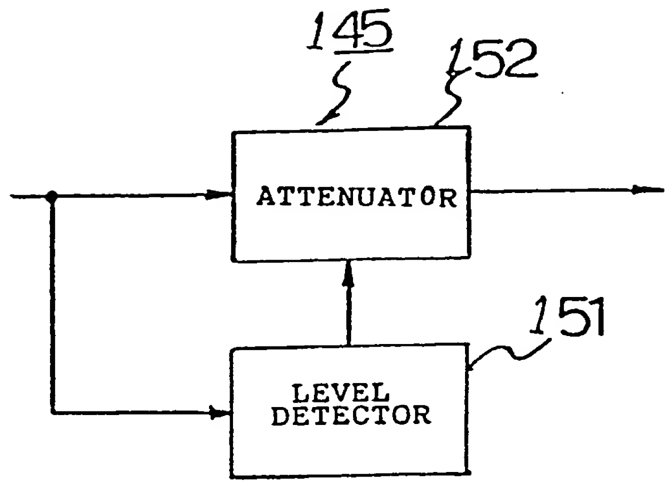


Fig. 4

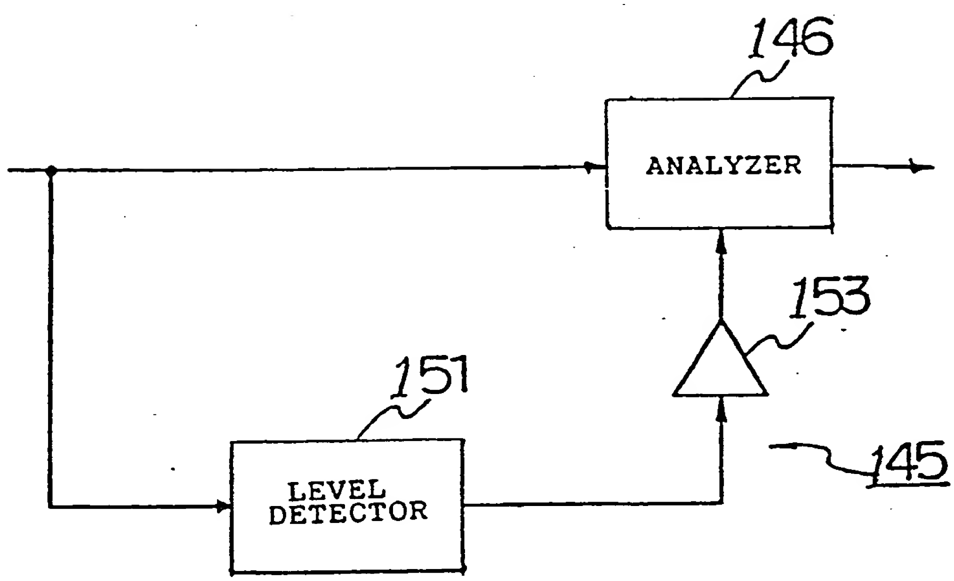


Fig. 5

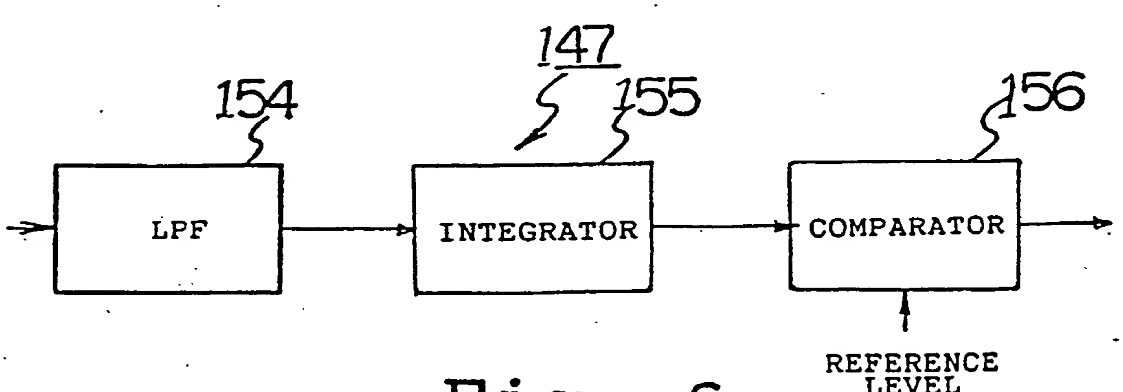


Fig. 6

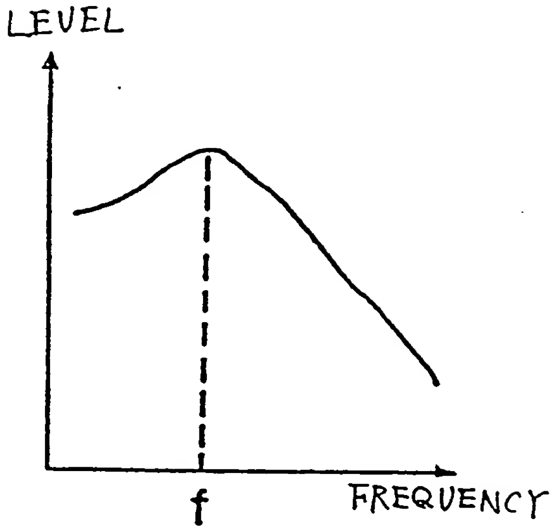


Fig. 7

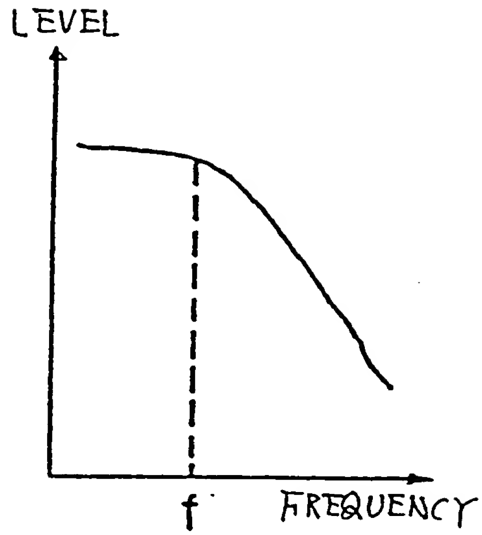


Fig. 8

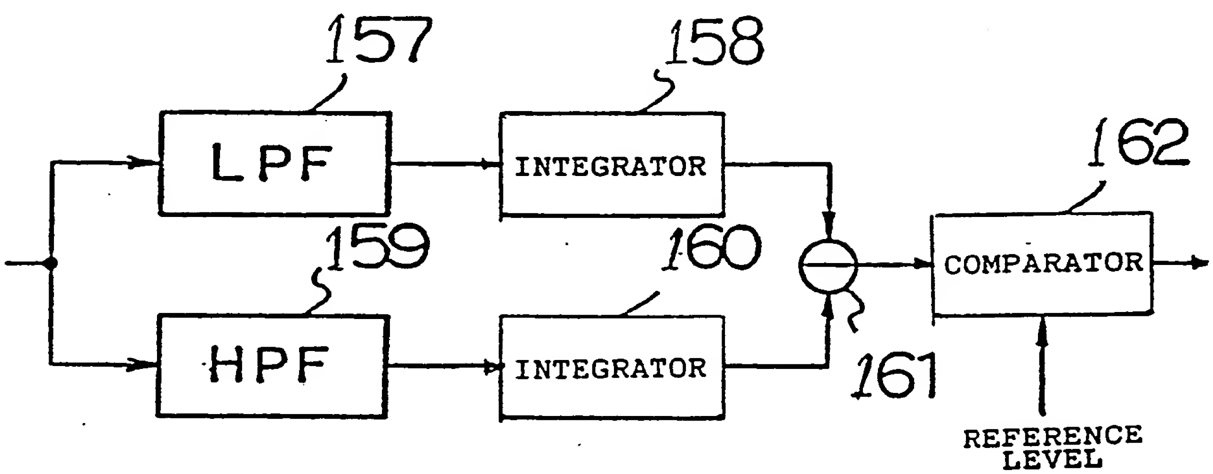
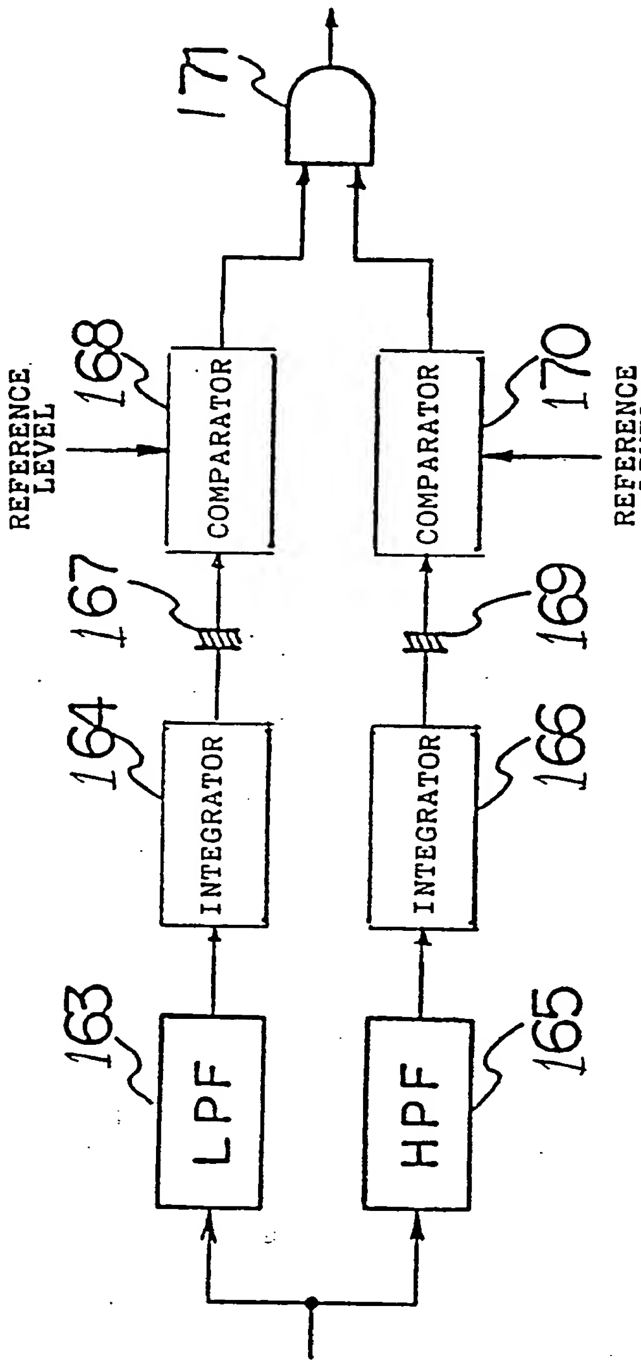


Fig. 9



LPF
HPF
INTEGRATOR
INTEGRATOR
COMPARATOR
COMPARATOR
REFERENCE LEVEL
REFERENCE LEVEL
163
164
165
166
167
168
169
170
171

Fig. 10

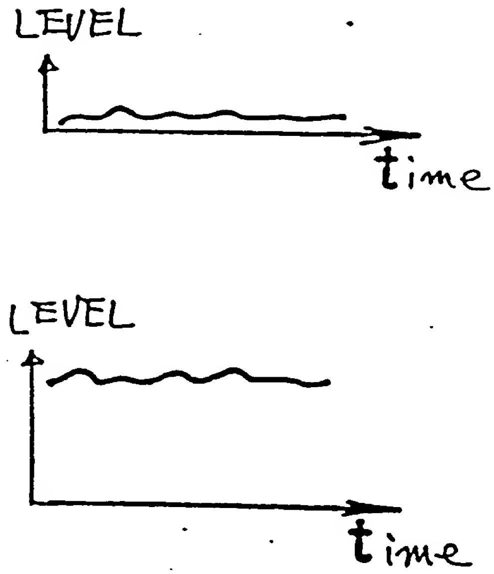


Fig. 11

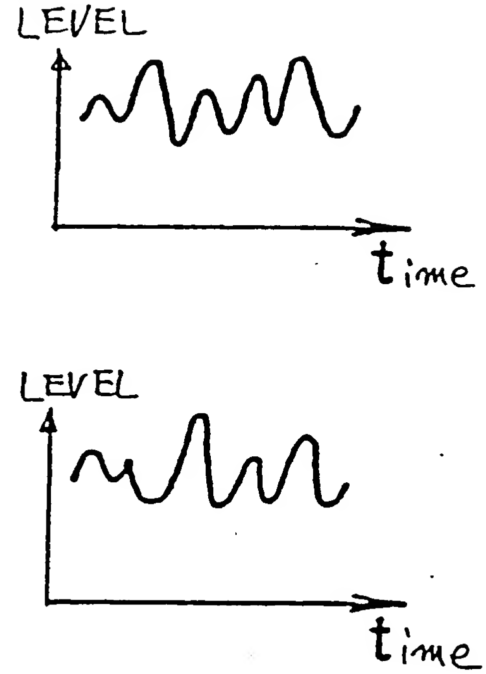


Fig. 12

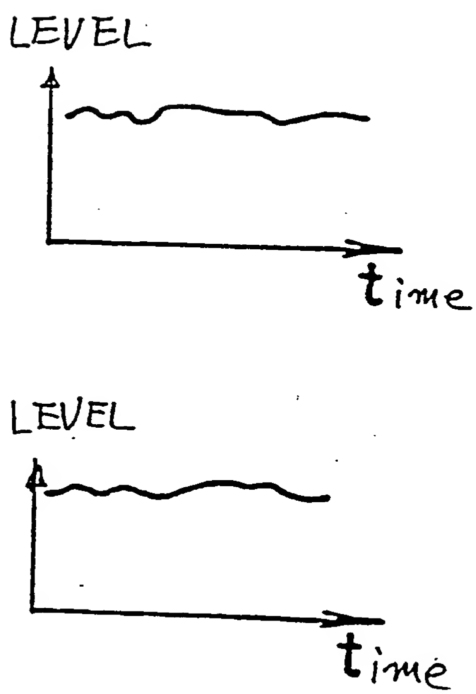


Fig. 13

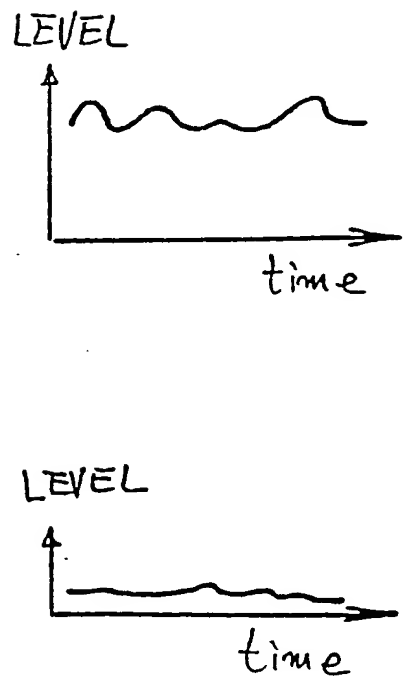


Fig. 14

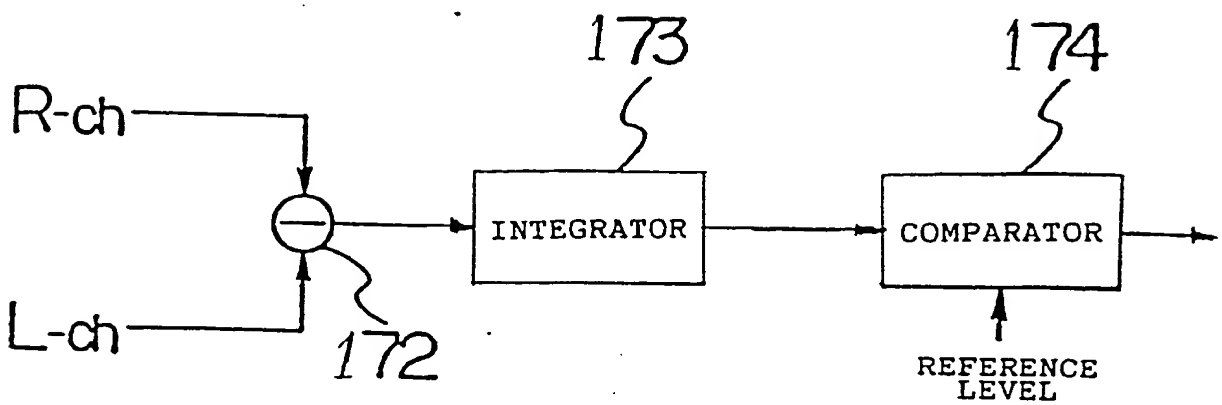


Fig. 15

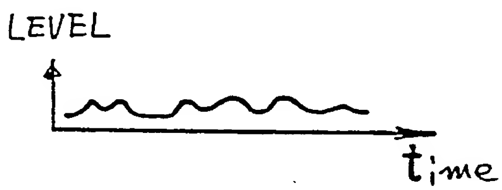


Fig. 16

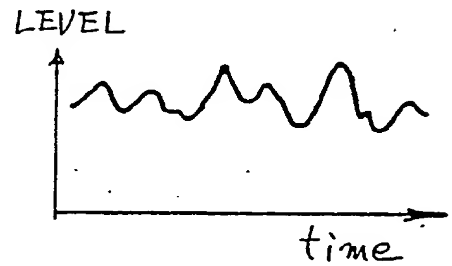


Fig. 17

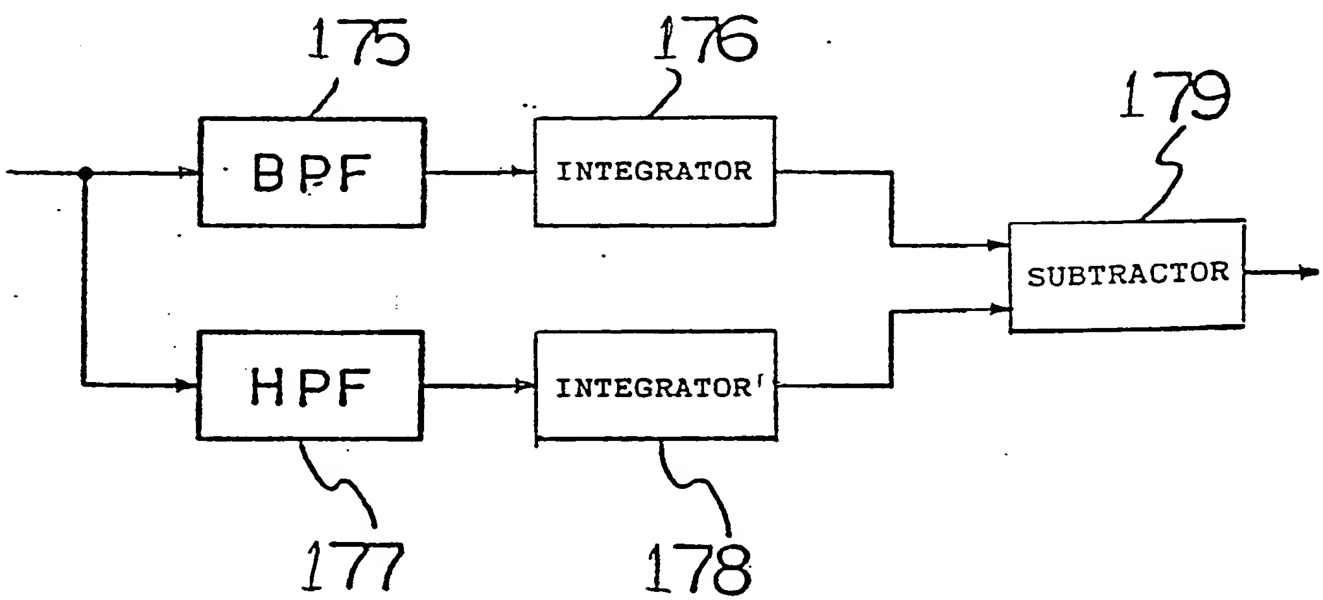


Fig. 18

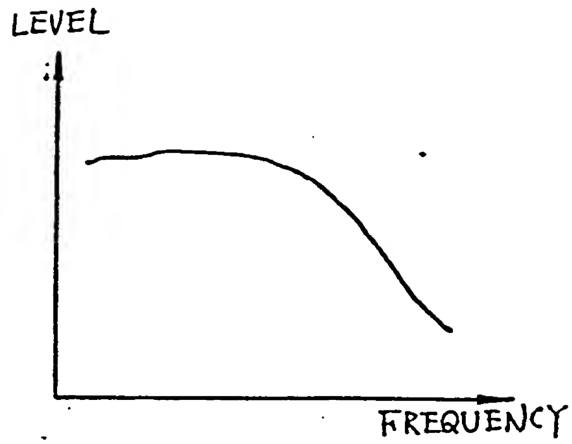


Fig. 19

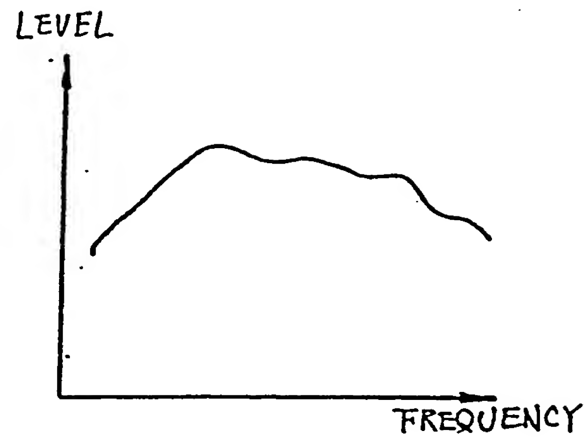


Fig. 20

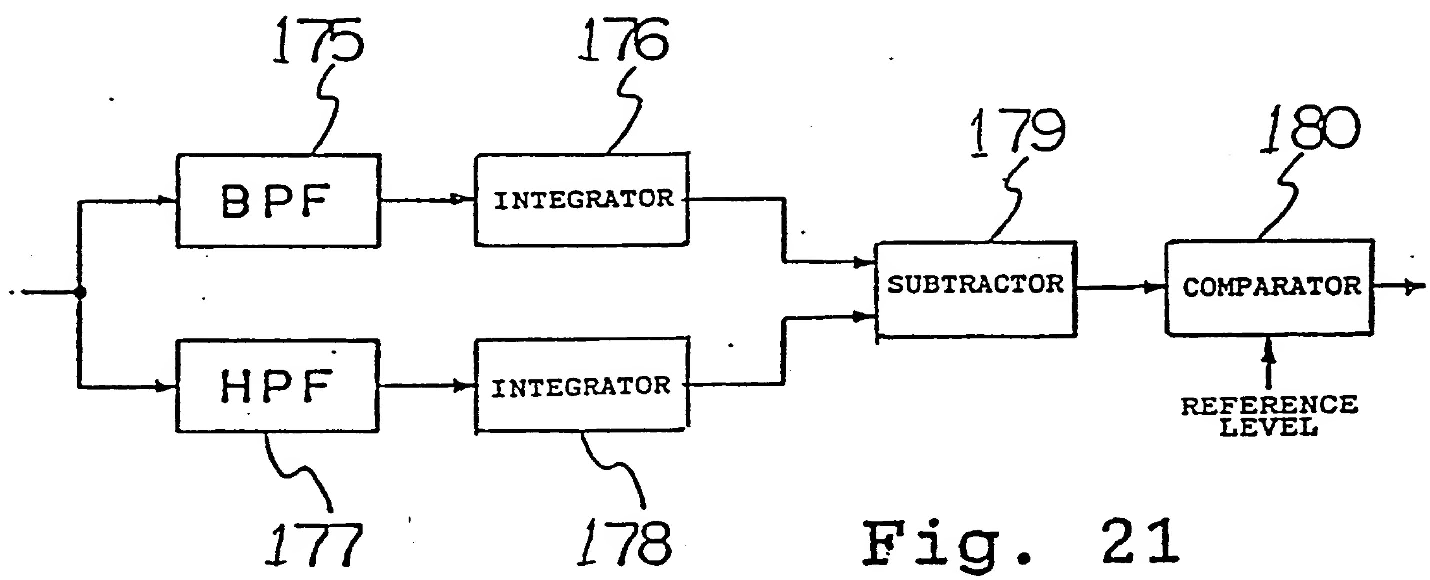


Fig. 21

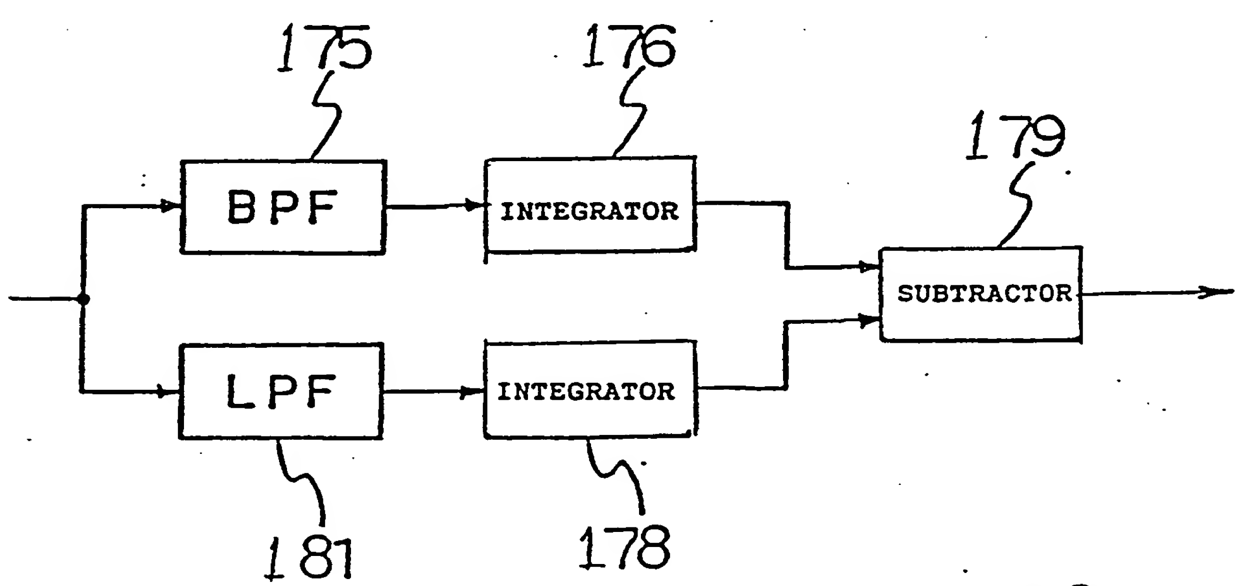


Fig. 22

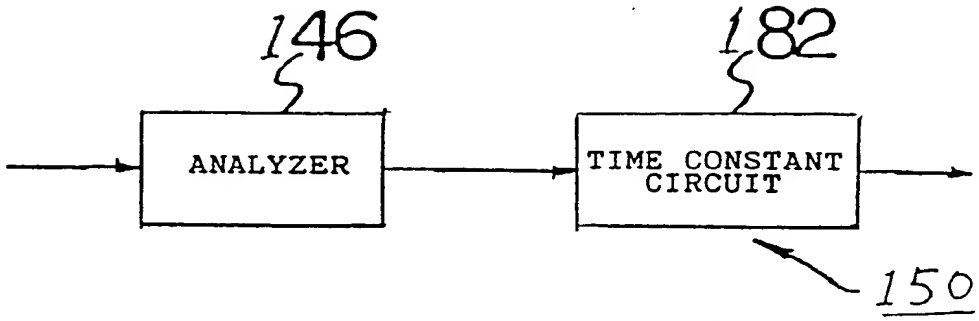


Fig. 23

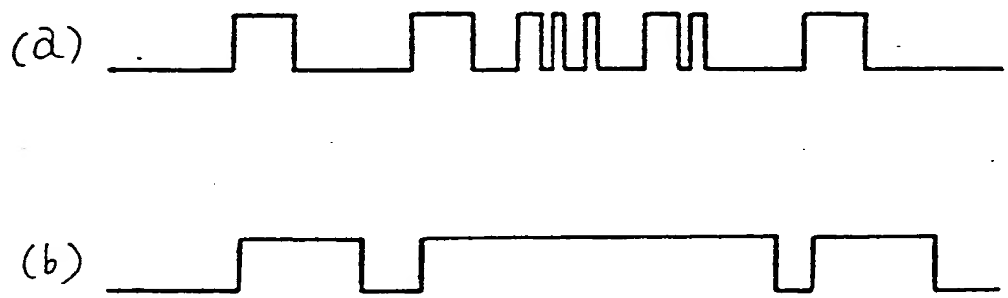


Fig. 24

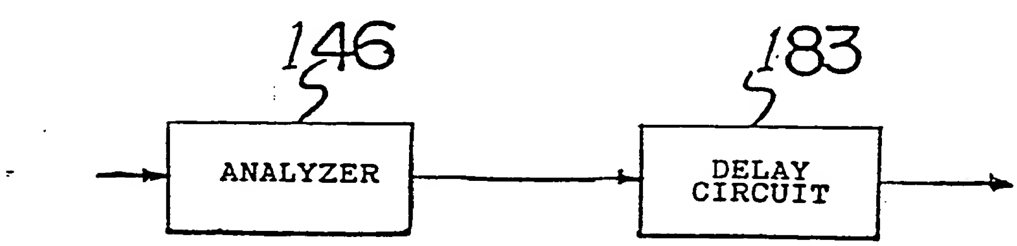


Fig. 25

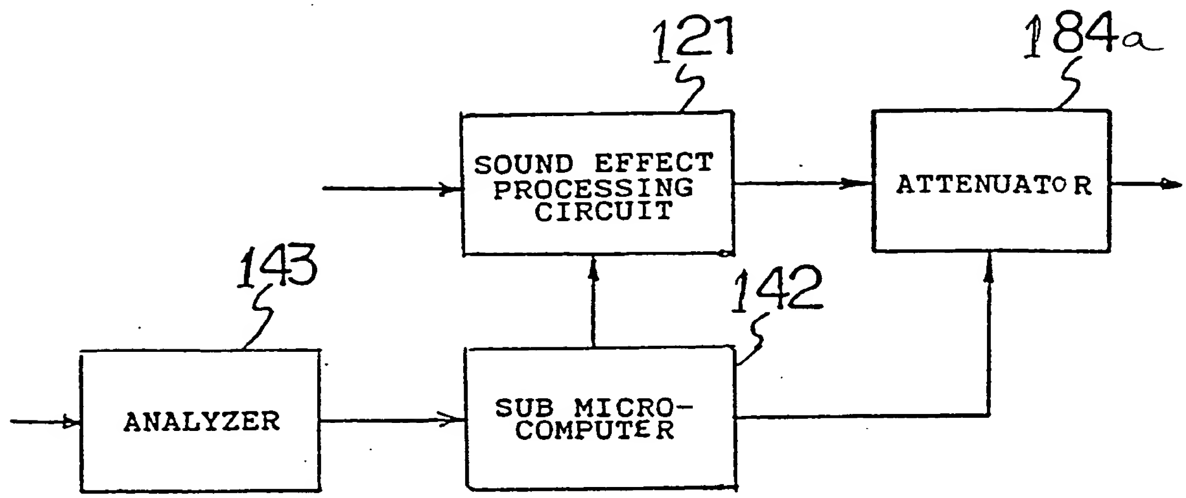

121
184a
SOUND EFFECT PROCESSING CIRCUIT
ATTENUATOR
143
142
ANALYZER
SUB MICRO-COMPUTER

Fig. 26

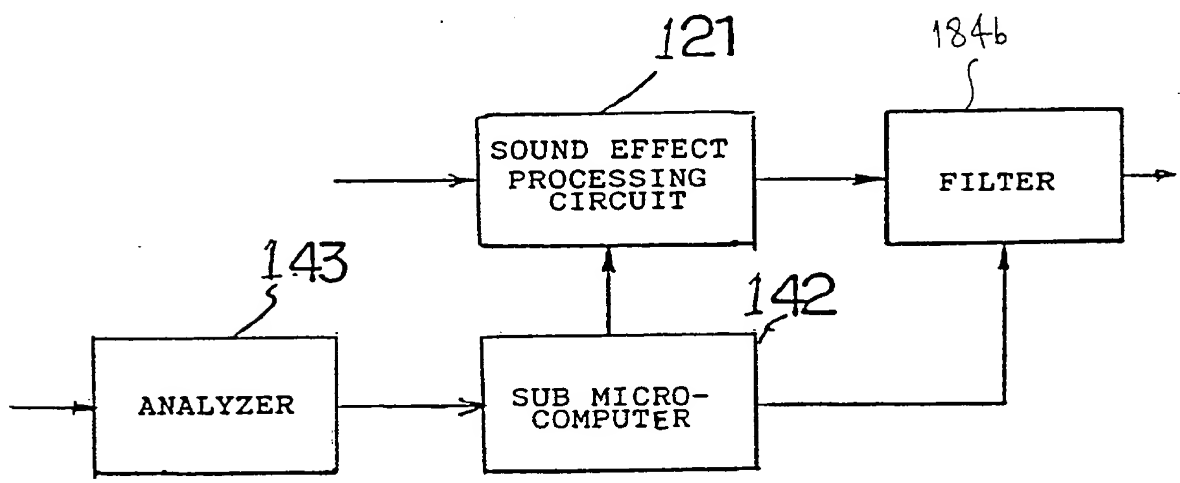

121
184b
SOUND EFFECT PROCESSING CIRCUIT
FILTER
143
142
ANALYZER
SUB MICRO-COMPUTER

Fig. 27

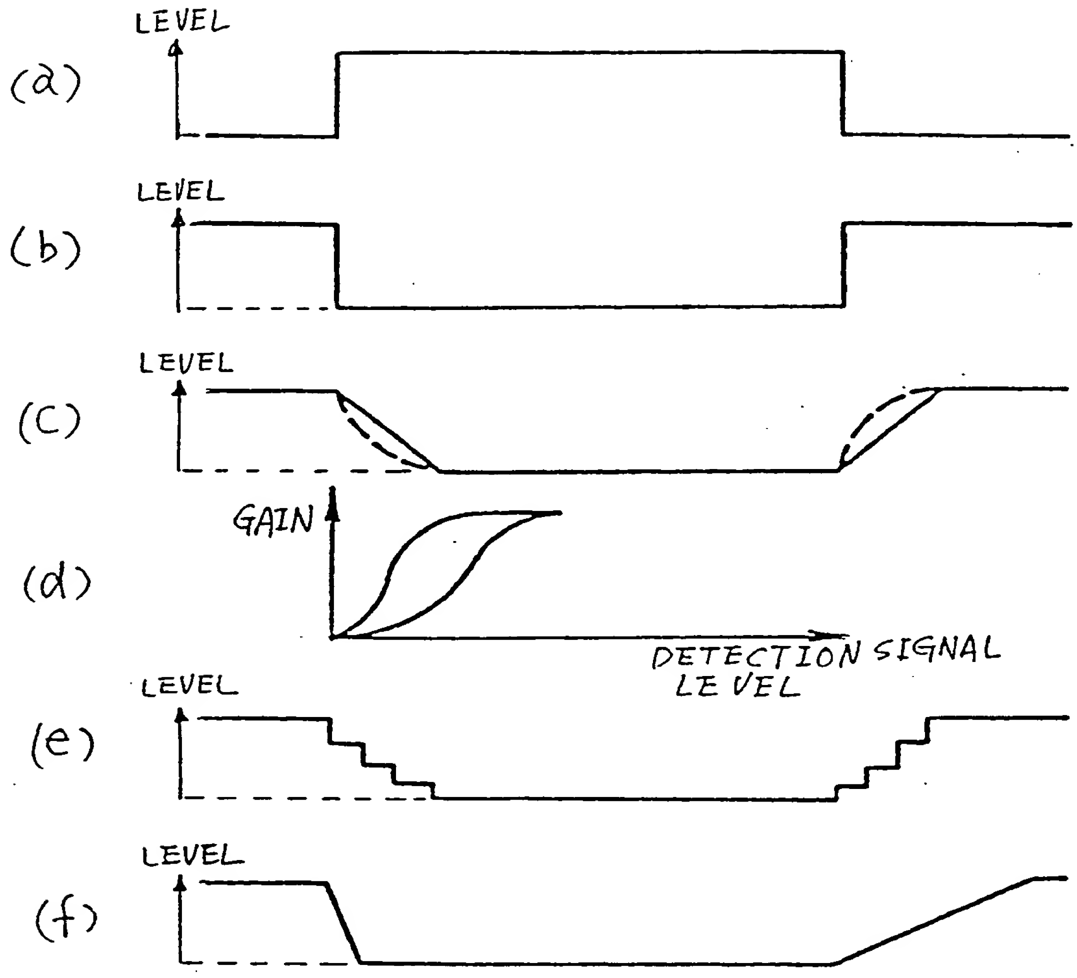

LEVEL
(a)
LEVEL
(b)
LEVEL
(c)
GAIN
(d)
DETECTION SIGNAL LEVEL
LEVEL
(e)
LEVEL
(f)

Fig. 28

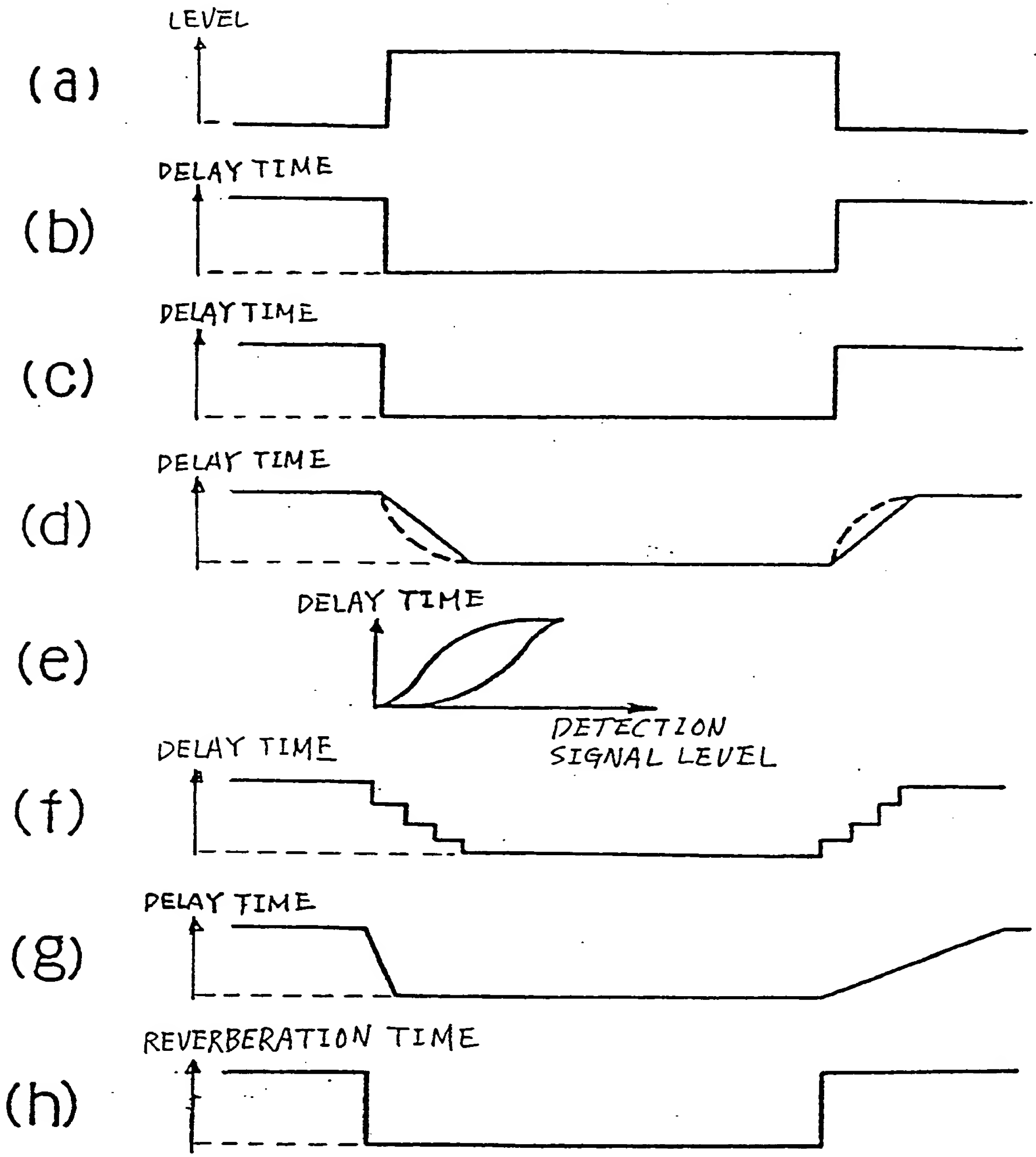

LEVEL
(a)
DELAY TIME
(b)
DELAY TIME
(c)
DELAY TIME
(d)
DELAY TIME
(e)
DETECTION SIGNAL LEVEL
DELAY TIME
(f)
DELAY TIME
(g)
REVERBERATION TIME
(h)

Fig. 29

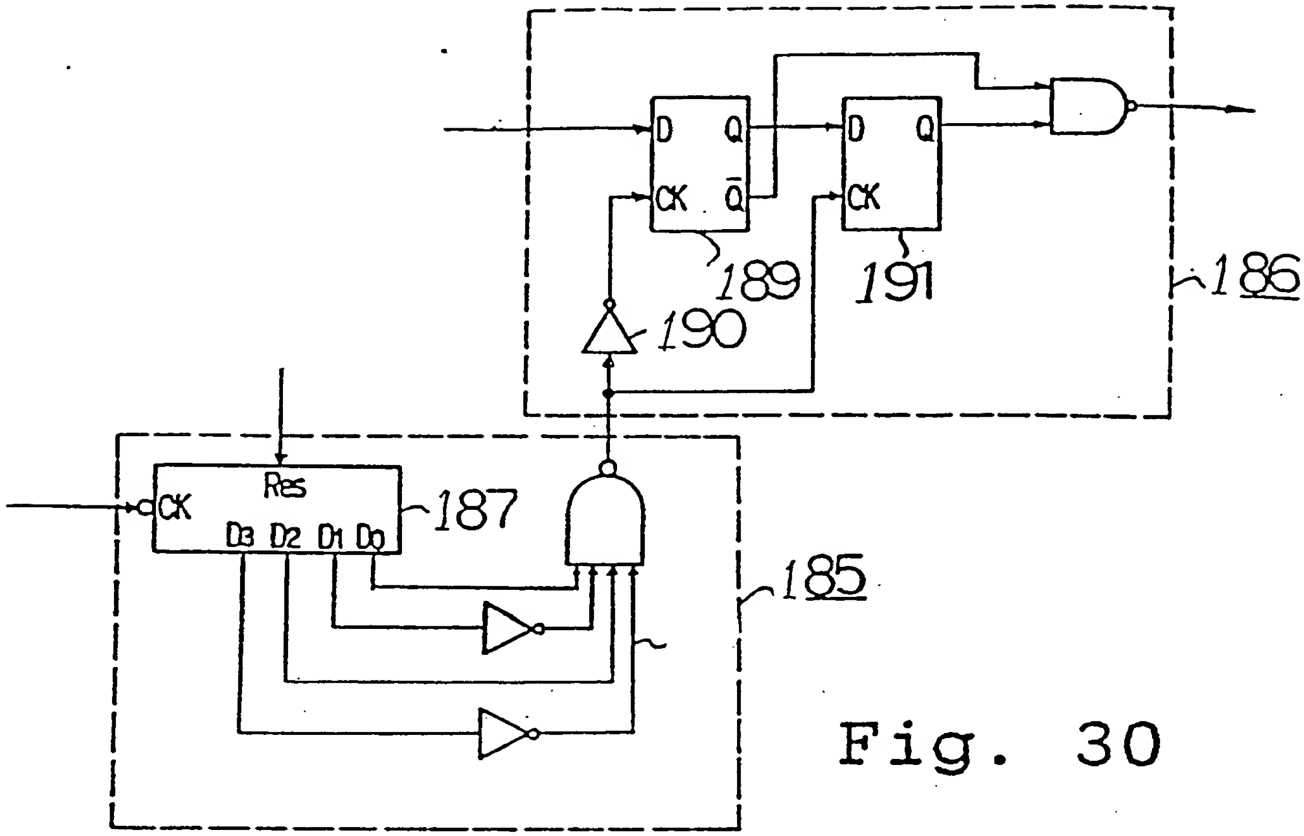

D
Q
CK
Q̄
D
Q
CK
189
191
186
190
Res
CK
D3
D2
D1
D0
187
185

Fig. 30

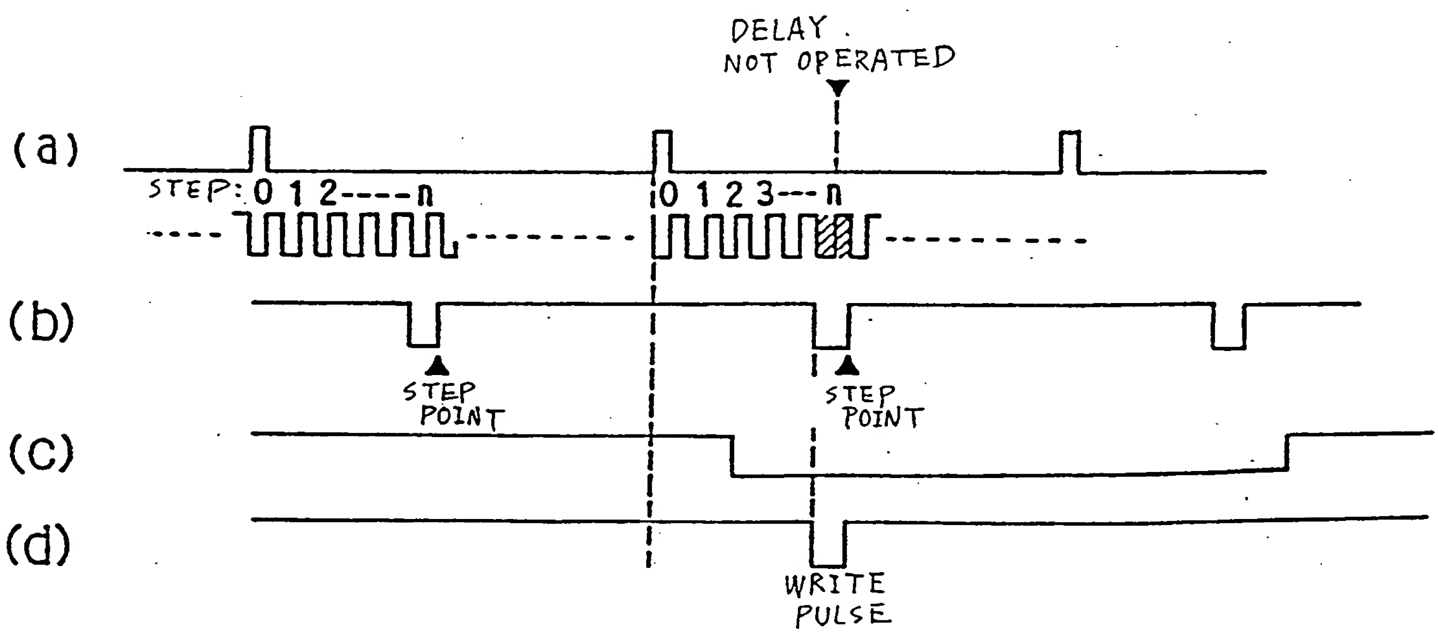

DELAY
NOT OPERATED
(a)
STEP: 0 1 2----n
0 1 2 3---n
(b)
STEP POINT
STEP POINT
(c)
(d)
WRITE PULSE

Fig. 31

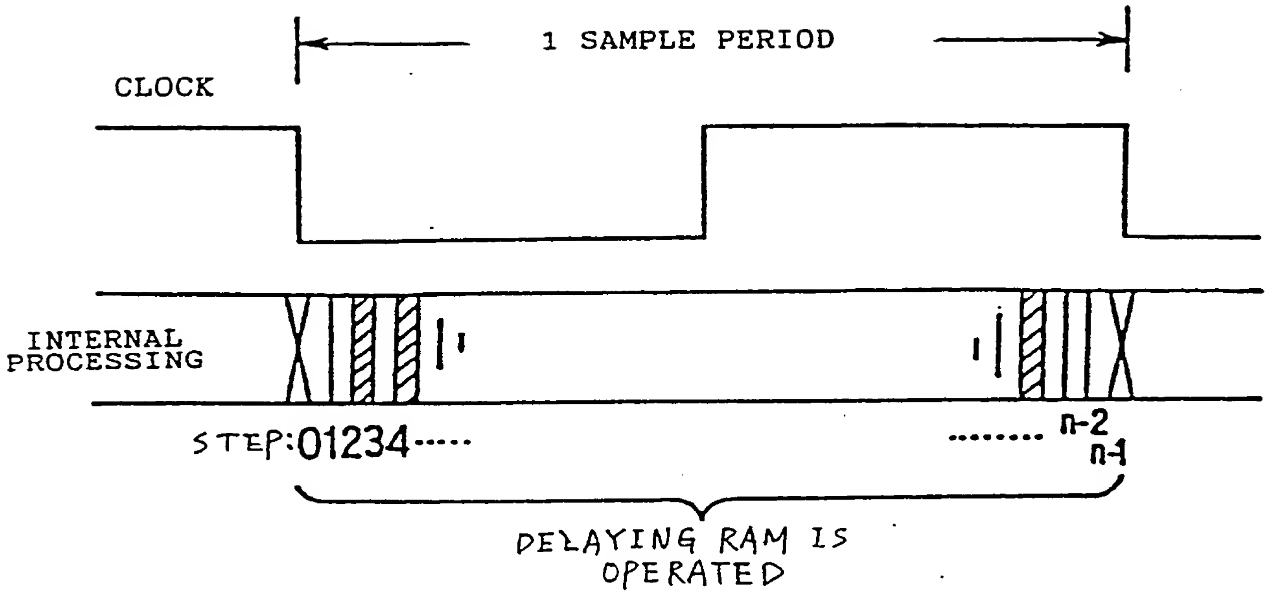

1 SAMPLE PERIOD
CLOCK
INTERNAL PROCESSING
STEP:01234·····
·······n-2
n-1
DELAYING RAM IS OPERATED

Fig. 32

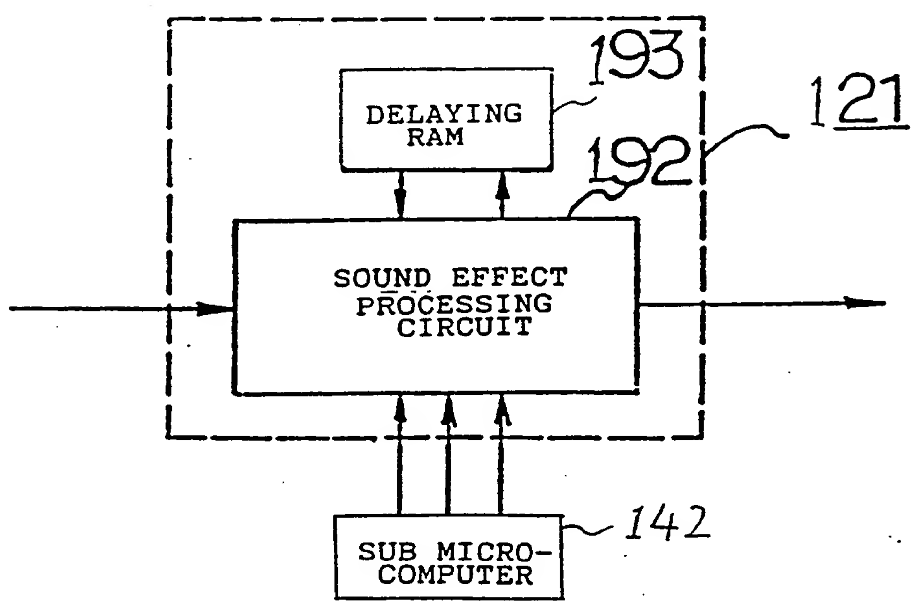

DELAYING RAM
193
121
192
SOUND EFFECT PROCESSING CIRCUIT
SUB MICRO-COMPUTER
142

Fig. 33

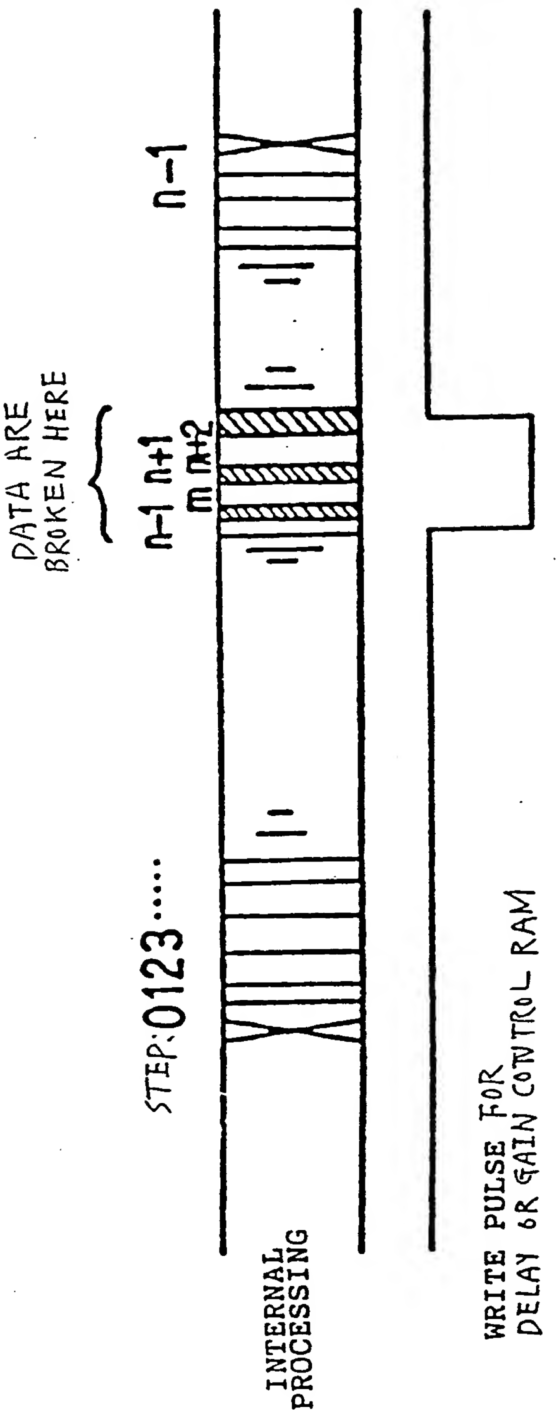

DATA ARE BROKEN HERE
n-1 n+1
m n+2
n-1
STEP:0123.....
INTERNAL PROCESSING
WRITE PULSE FOR DELAY OR GAIN CONTROL RAM

Fig. 34

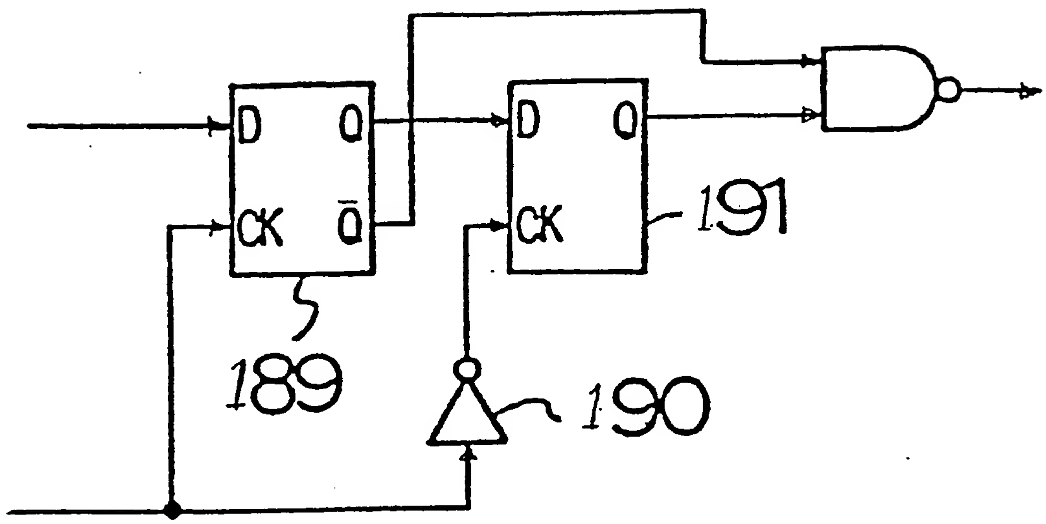


Fig. 35

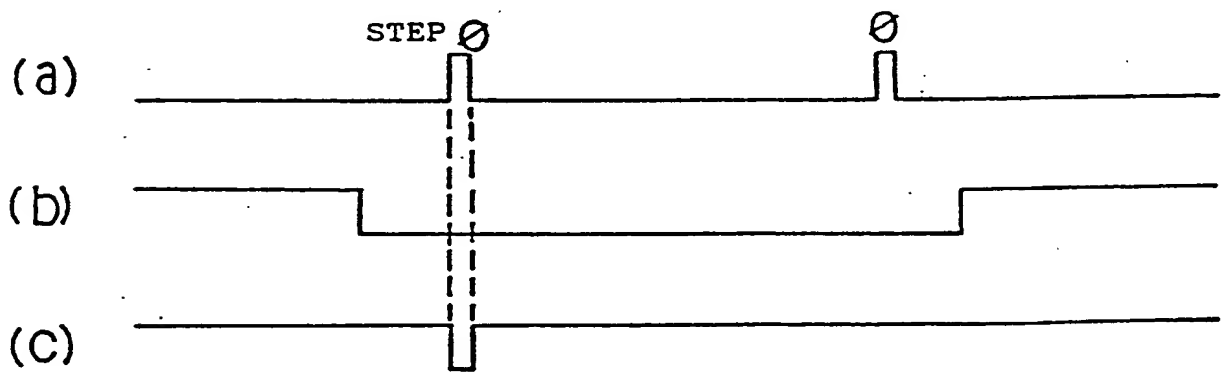


Fig. 36

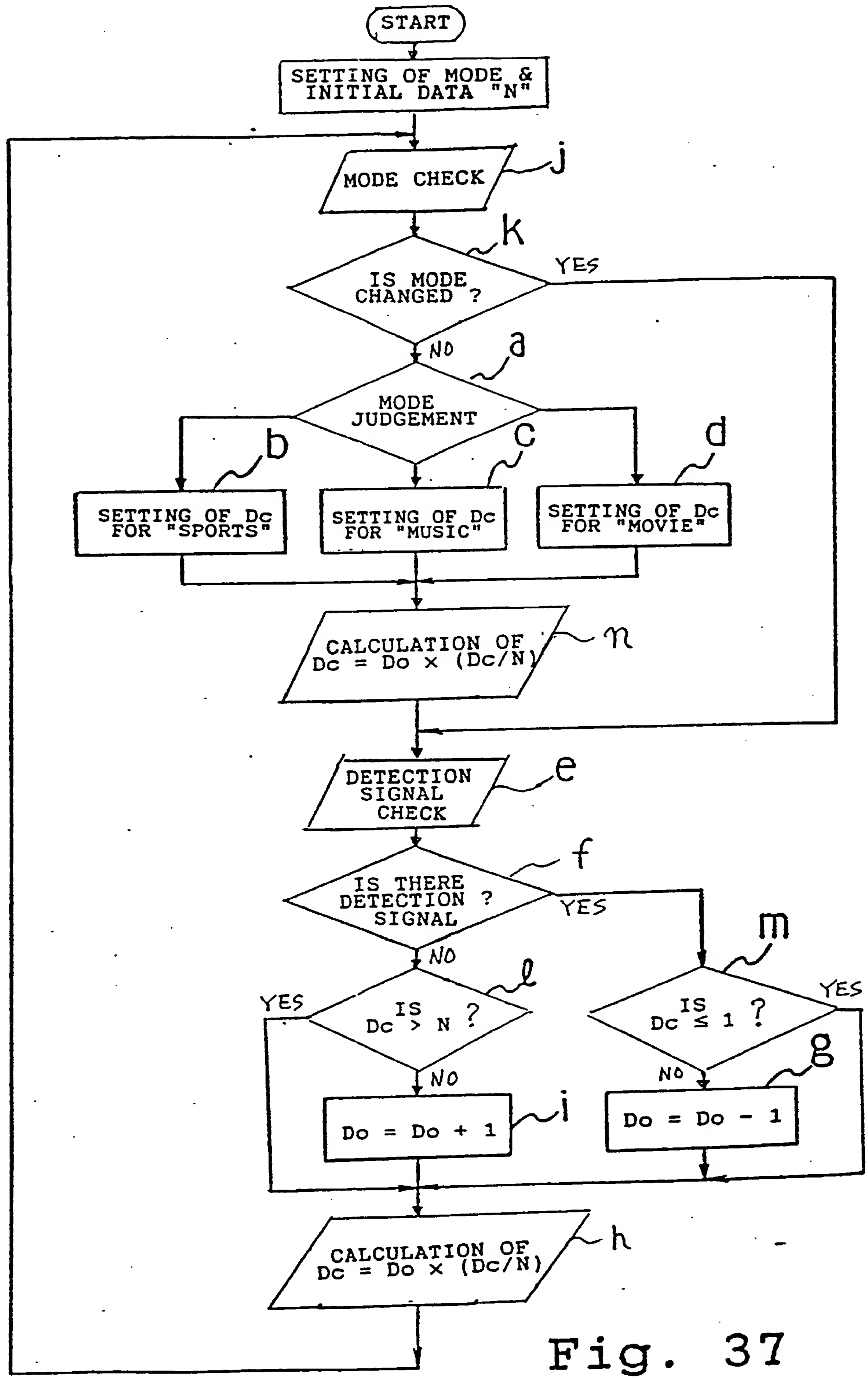

START
SETTING OF MODE & INITIAL DATA "N"
MODE CHECK
j
IS MODE CHANGED ?
k
YES
NO
MODE JUDGEMENT
a
SETTING OF Dc FOR "SPORTS"
b
SETTING OF Dc FOR "MUSIC"
c
SETTING OF Dc FOR "MOVIE"
d
CALCULATION OF Dc = Do x (Dc/N)
n
DETECTION SIGNAL CHECK
e
IS THERE DETECTION SIGNAL ?
f
YES
NO
IS Dc > N ?
l
YES
NO
Do = Do + 1
i
IS Dc ≤ 1 ?
m
YES
NO
Do = Do - 1
g
CALCULATION OF Dc = Do x (Dc/N)
h

Fig. 37

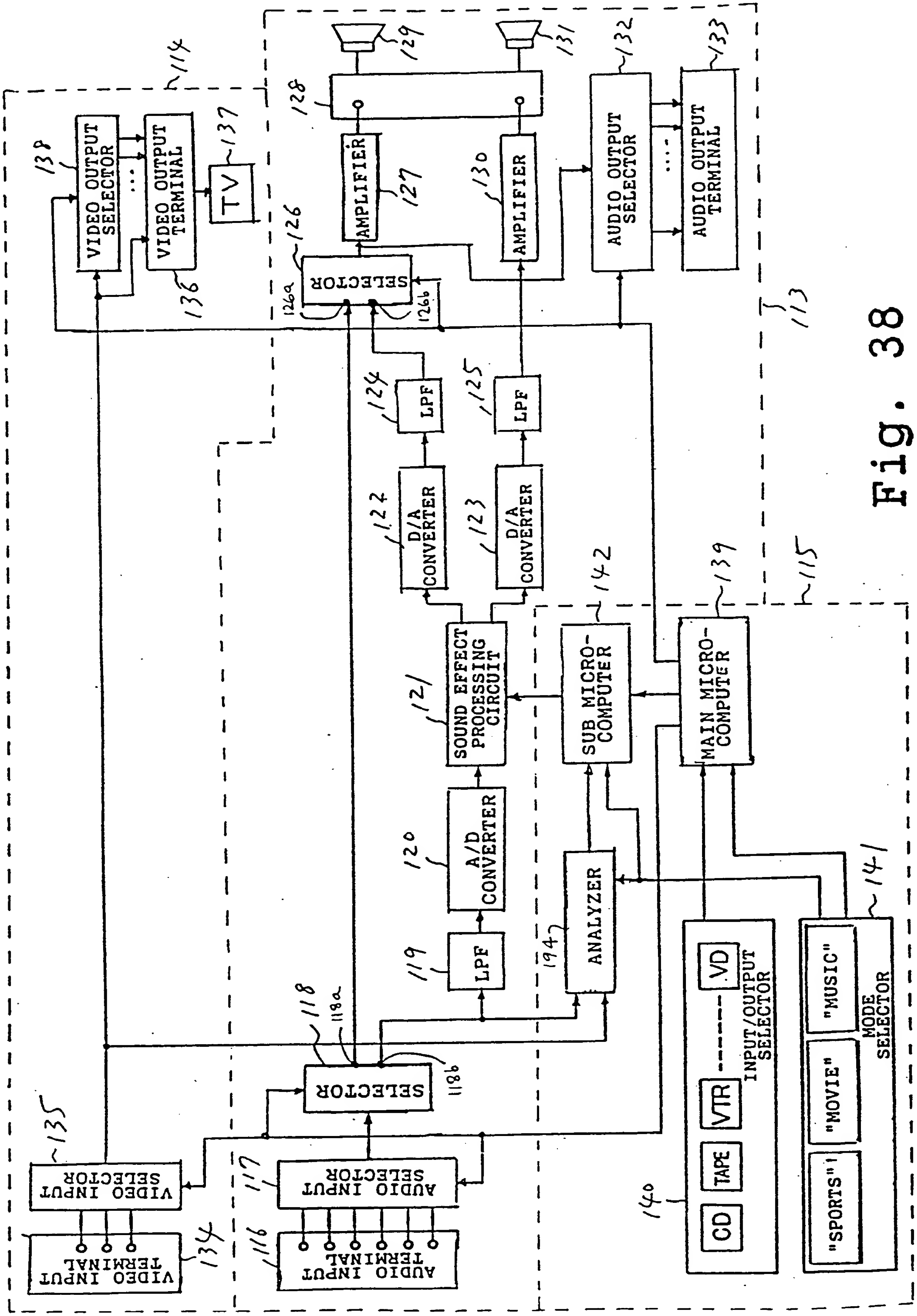

VIDEO OUTPUT SELECTOR
VIDEO OUTPUT TERMINAL
TV
AMPLIFIER
AMPLIFIER
AUDIO OUTPUT SELECTOR
AUDIO OUTPUT TERMINAL
SELECTOR
LPF
LPF
D/A CONVERTER
D/A CONVERTER
SOUND EFFECT PROCESSING CIRCUIT
SUB MICRO-COMPUTER
MAIN MICRO-COMPUTER
A/D CONVERTER
LPF
ANALYZER
SELECTOR
VIDEO INPUT SELECTOR
AUDIO INPUT SELECTOR
VIDEO INPUT TERMINAL
AUDIO INPUT TERMINAL
CD
TAPE
VTR
VD
INPUT/OUTPUT SELECTOR
"SPORTS"
"MOVIE"
"MUSIC"
MODE SELECTOR
113
114
115
116
117
118
118a
118b
119
120
121
122
123
124
125
126
126a
126b
127
128
129
130
131
132
133
134
135
136
137
138
139
140
141
142
194

Fig. 38

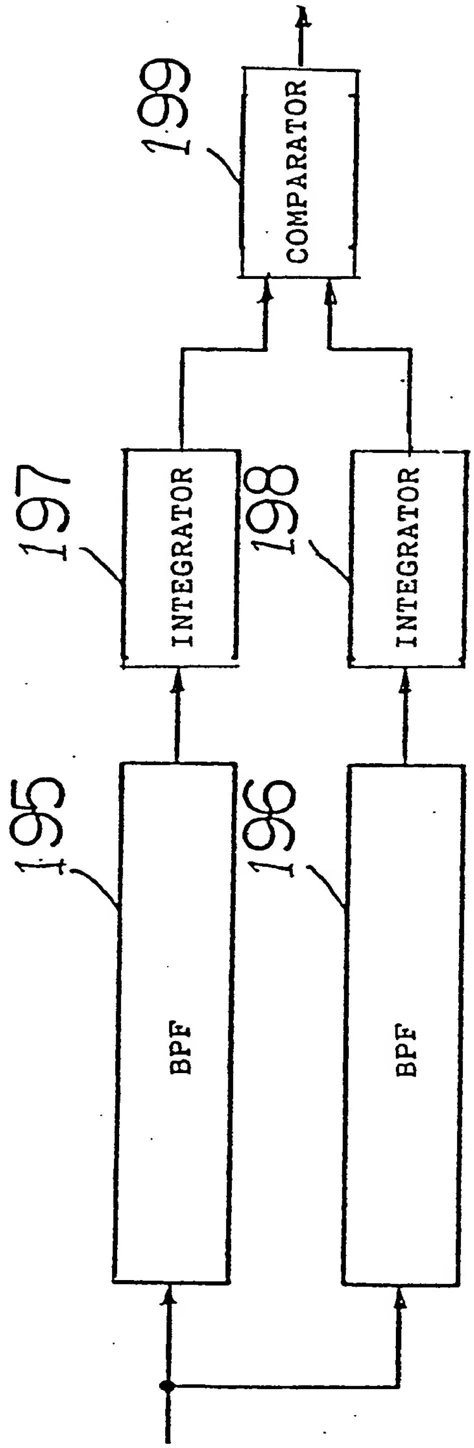

199
COMPARATOR
197
INTEGRATOR
198
INTEGRATOR
195
BPF
196
BPF

Fig. 39

(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) Publication number: 0 367 569 A3

(12) EUROPEAN PATENT APPLICATION

(21) Application number: 89311250.8

(22) Date of filing: 31.10.89

(30) Priority: 31.10.88 JP 274726/88

(43) Date of publication of application: 09.05.90 Bulletin 90/19

(84) Designated Contracting States: DE FR GB

(88) Date of deferred publication of the search report: 24.07.91 Bulletin 91/30

(51) Int. Cl.5: H04S 1/00, H04R 27/00

(71) Applicant: KABUSHIKI KAISHA TOSHIBA
72, Horikawa-Cho Saiwai-ku
Kawasaki-shi Kanagawa-ken(JP)

Applicant: TOSHIBA AUDIO VIDEO ENGINEERING CO. LTD.
3-9, Shinbashi 3-chome Minato-ku
Tokyo105(JP)

(72) Inventor: Sasaki, Akira Intellectual Property Division
Toshiba Corporation 1-1-1, Shinbaura
Minato-ku
Tokyo(JP)
Inventor: Suzuki, Katsuyoshi Intellectual Property Division
Toshiba Corporation 1-1-1, Shinbaura
Minato-ku
Tokyo(JP)
Inventor: Sakai, Kazuyasu Intellectual Property Division
Toshiba Corporation 1-1-1, Shinbaura
Minato-ku
Tokyo(JP)

(74) Representative: Muir, Ian R. et al
HASELTINE LAKE & CO. Hazlitt House 28
Southampton Buildings Chancery Lane
London WC2A 1AT(GB)

(54) Sound effect system.

(57) An audio signal processing apparatus for processing audio signal including an audio signal input circuit (116 -118) into which the audio signals are input, an audio signal analyzer (143) which analyzes the input audio signals and generates an output control signal, a sound effect processor (121) which performs a prescribed sound effect processing on the input audio signals and outputs a resulting audio signal, a controller (142) which controls the sound effect processor (121) to optimize the sound effect processing in response to the control signal from the audio signal analyzer (143) and an audio signal output circuit (122 -131) for outputting the resulting audio signal.

EP 0 367 569 A3

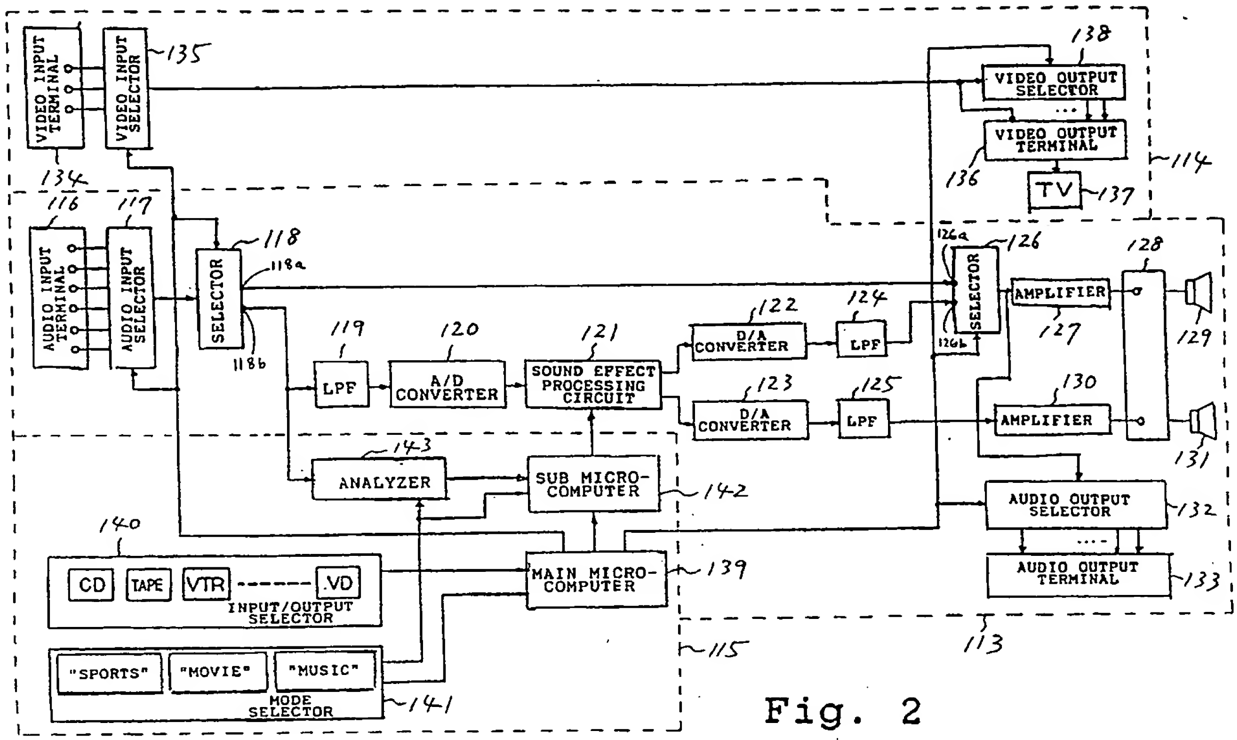

VIDEO INPUT TERMINAL
VIDEO INPUT SELECTOR
135
134
AUDIO INPUT TERMINAL
AUDIO INPUT SELECTOR
116
117
SELECTOR
118
118a
118b
LPF
119
A/D CONVERTER
120
SOUND EFFECT PROCESSING CIRCUIT
121
D/A CONVERTER
122
D/A CONVERTER
123
LPF
124
LPF
125
SELECTOR
126
126a
126b
AMPLIFIER
127
128
129
AMPLIFIER
130
131
AUDIO OUTPUT SELECTOR
132
AUDIO OUTPUT TERMINAL
133
113
VIDEO OUTPUT SELECTOR
138
VIDEO OUTPUT TERMINAL
136
TV
137
114
ANALYZER
143
SUB MICRO-COMPUTER
142
MAIN MICRO-COMPUTER
139
CD
TAPE
VTR
VD
INPUT/OUTPUT SELECTOR
140
"SPORTS"
"MOVIE"
"MUSIC"
MODE SELECTOR
141
115

Fig. 2

European
Patent Office

EUROPEAN SEARCH REPORT

Application Number

EP 89 31 1250

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int. Cl.5)
X Y	US-A-4 694 497 (KASAI et al.) * The whole document *	1-18 19-21	H 04 S 1/00 H 04 R 27/00
X Y A	EP-A-0 276 948 (YAMAHA) * The whole document *	1 19-21 7-18	
A	US-A-4 856 064 (IWAMATSU) * The whole document *	7-18	

TECHNICAL FIELDS SEARCHED (Int. Cl.5)

H 04 R
H 04 S
G 10 H

The present search report has been drawn up for all claims

Place of search	Date of completion of search	Examiner
The Hague	25 April 91	BEHRINGER L.V.

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same catagory
A : technological background
O : non-written disclosure
P : intermediate document
T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document